IDA8 & Ateis Studio User Manual

Version 1.0.1

DELIVERING YOUR MESSAGE!

Revision History

Time	Version	
2012/07/23	1.0.1	Modified DNM chapter in the Product Features> Consoles and Accessories.
2012/07/11	1.0.0	The First Version.

Table of Contents

1 Introduction

1.1 Welcome

Thank you for choosing ATEÏS. We here at ATEÏS hope you will enjoy our technology as much as we enjoyed developing and building it.

This manual is intended to provide the user with the necessary understanding of our system architecture as well as guide users through the configuration process.

This manual can be updated at any time without prior notice in order to keep it up to date.

If you find errors in this manual or would like to improve on the presentation, feel free to submit mistakes, suggestions or questions by sending an email.

We hope that this Help File Manual will provide you all the information you need. However if you have any questions, feel free to contact us.

1.2 Ateis Presentation

ATEÏS has been in the professional audio market for close to thirty years and is viewed as a leading competitor in the Public Address, Voice Alarm, and Professional Audio Market in Europe, Asia, and the Middle East.

- Products

  The company offers a full range of audio equipment: microphones, preamplifiers, digital processors, digital audio matrixes, loud-speaker monitoring systems, amplifiers, etc. ATEÏS designs and manufactures leading products in the voice alarm systems market which have been certified EN60849 compliant by the TÜV and UL listed.

- Development

  Thanks to a development team of over forty engineers and a close connection to our customer base, we are able to respond rapidly to the demands of our various vertical markets with specific solutions and cutting edge technology. You can rest assured that our technology is always cutting edge with a view to the future.

- ATEÏS Vertical Markets

  - Transportation (Railways, Subways, Airports)
  - High rise buildings

- Hotels
- Restaurants
- Shopping malls
- Theme parks
- Places of worship
- Stadia
- Museums
- Industrial
- University and campus applications

1.3 EC Declaration of Conformity

Documentation CE marking. This needs to be added to each commercial/technical document

CE

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EC DECLARATION OF CONFORMITY

Ateis, as manufacturer having sole responsibility, hereby declares that our delivered version of the following product complies with the applicable directives. In case of alterations of the product, not agreed upon by Ateis, this declaration is no longer valid.

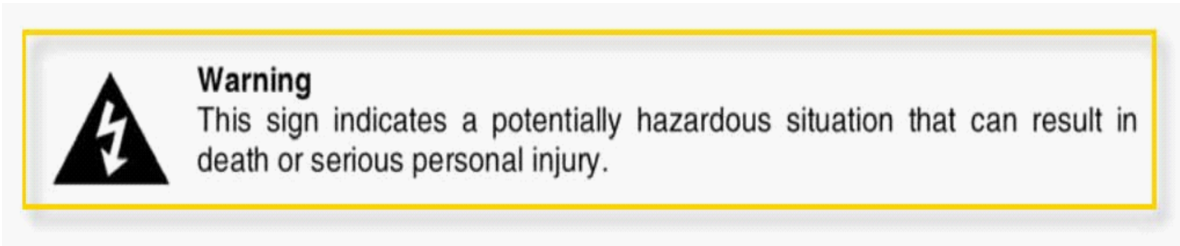
EN54-16:2008
Voice Alarm Control and Indicating Equipment (VACIE)

Product: **IDA8 System VACIE**

Certified options:
Audible warning
Manual reset of the voice alarm condition
Voice alarm condition output
Indication of faults related to the transmission path of the CIE
Indication of faults related to voice alarm zones
Voice alarm manual control
Interface to external control device(s)
Emergency microphones
Redundant amplifiers

Ancillary functions:
Commercial paging
Backgroud music distribution
Audio and data networking

2 Safety Declartion



❖ Do not expose the device to extreme temperatures, direct sunlight, humidity, or dust, which could cause fire or electrical shock hazard.

❖ Keep away water or other liquids from the device. Otherwise fire or electrical shock may result.

❖ Connect the power cord only to an AC outlet of the type stated in this Owner's Manual or as marked on the unit. Otherwise fire and electrical shock hazard results.

❖ When disconnecting the power cord from an AC outlet always grab the plug. Never pull the cord. A damaged power cord is a potential risk of fire and electrical shock hazard.

❖ Avoid touching power plugs with wet hands. Doing so is a potential electrical shock hazard.

❖ Take care for correct polarity when operating the device from a DC power source. Reversed polarity may cause damage to the unit or the batteries.

❖ Avoid placing heavy objects on power cords. A damaged power cord is a fire and electrical shock hazard.

❖ Do not cut, scratch, bend, twist, pull, or heat the power cord. A damaged power cord is a fire and electrical shock hazard. Ask your ATEÏS dealer for replacement.

❖ Turn off immediately the unit, remove the power cord from the AC outlet and consult your ATEÏS dealer in any of the following circumstances:

- Smoke, odor, or noise getting out of the unit.
- Foreign objects or liquids get inside the device.
- The unit has been dropped or the shell is damaged.
- The power cord is damaged.

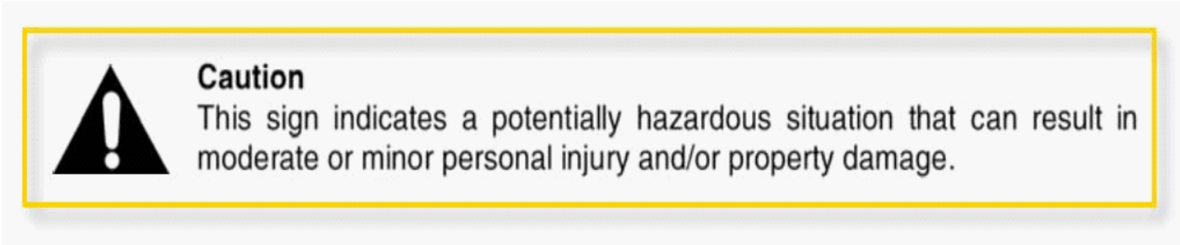
If you continue using the device, fire and electrical shock may result.

❖ Do not drop or insert metallic objects or flammable materials into the unit as this may result in fire and electrical shock.

❖ Do not remove the device's cover, as there are exposed parts inside carrying high voltages that may cause an electrical shock. Contact your ATEÏS dealer if internal inspection, maintenance, or repair is necessary.

❖ Do not try to make any modifications to the device. This is a potential fire and electrical shock hazard.

❖ Avoid the device's ventilation slots to be blocked. Blocking the ventilation slots is a potential fire hazard.



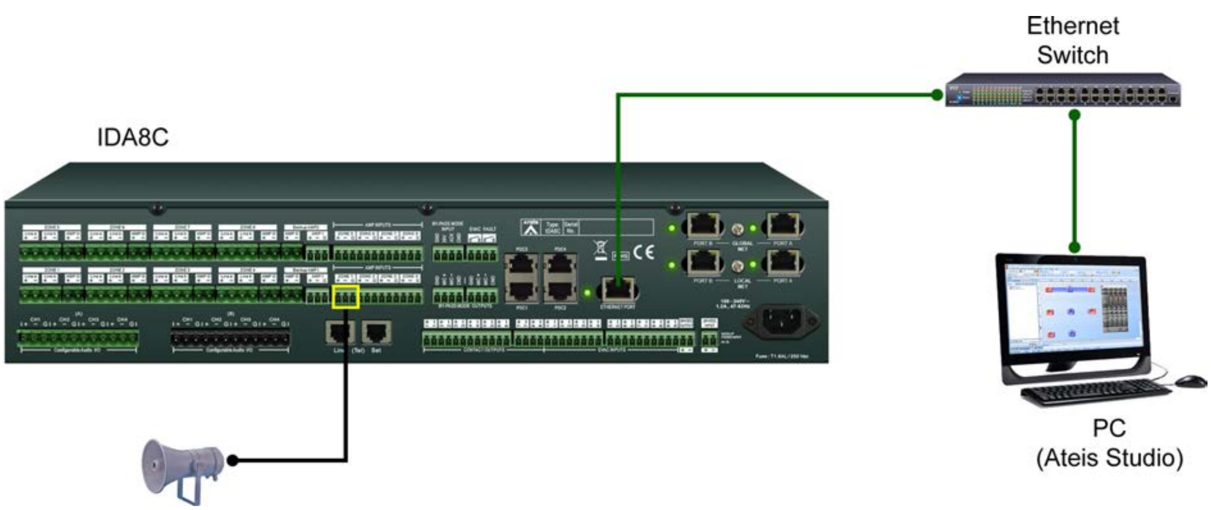
- To prevent the unit from falling down and causing personal injury and/or property damage, avoid installing or mounting the unit in unstable locations.
- Leave enough space above and below the unit to provide good ventilation of the device. If the airflow is not adequate, the device will heat up inside and may cause a fire.
- Operate the device in an environment with a free-air temperature of between 0 °C and 40 °C (32 °F and 104 °F).
- Turn off all audio equipment when making any connections to the device, and make sure to use adequate cables.
- Do not use benzene, thinner, or chemicals to clean the device. Use only a soft, dry cloth.
- If the device is moved from a cold place (e.g., overnight in a car) to a warmer environment, condensation may form inside the unit, which may affect performance. Allow the device to acclimatize for about one hour before use.

3 Quick Start

3.1 for IDA8

Here is a simple demonstration showing how to configure one of our platforms and how to adjust parameters to get a 2K tone from the connected speaker. This example uses an IDA8C, but the other audio processors follow the same procedure.

1. Setup device and wiring



- Speaker is connect to Amplifier Zone Output(Zone1).

- Connect ethernet wire for device and PC.IDA8C.

2. Power up device

   Power up device and make sure power LED is on.

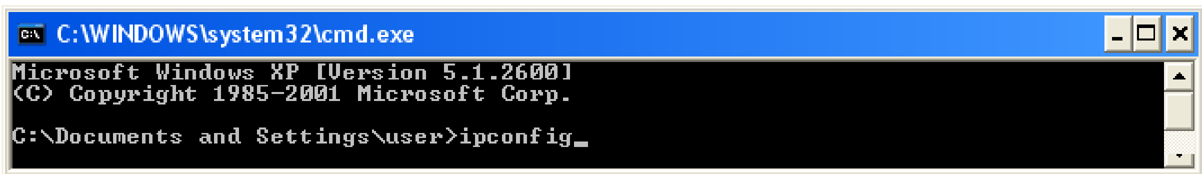
3. Set IP, Subnet mask and gateway address of device

   To set IP, Subnet mask and Gateway address, use the menu on the touch screen on the front of the IDA8C. If you are using a platform that does not have a touch screen, use the other method to setup, see the related topics of it. In this case, set IP = 192.168.100.79.

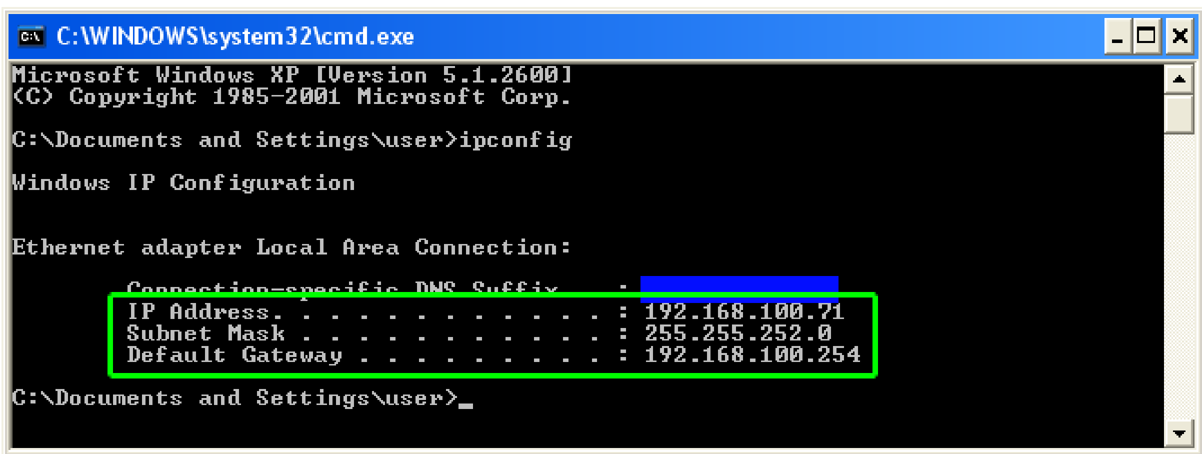
   The subnet mask and geteway address should be the same as computer's.

   You can find information from cmd window:

   Type "ipconfig" in cmd window:

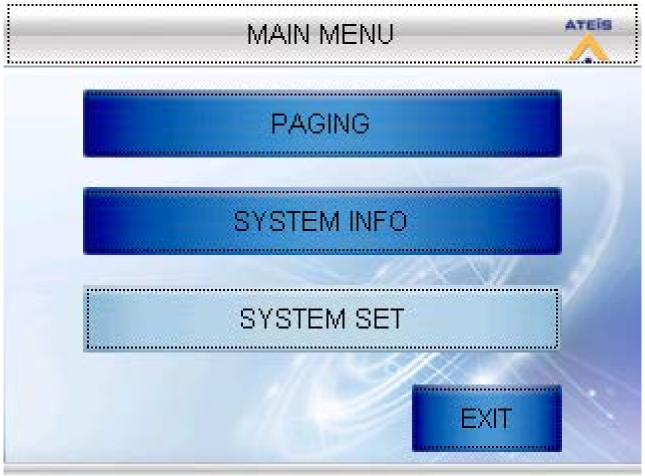


   After execute command "ipconfig" IP/Subnet Mask/Gateway information is displayed in window, inside the green rectangle:

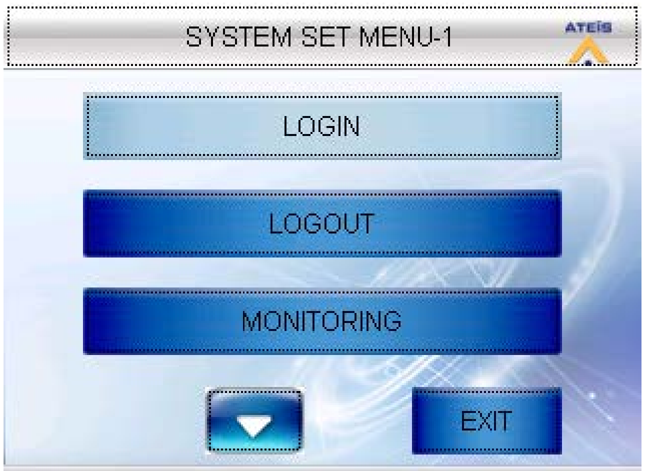


   In IDA8C, IP/Subnet Mask/Gateway allow modify if user login the system. Follow the steps to login:

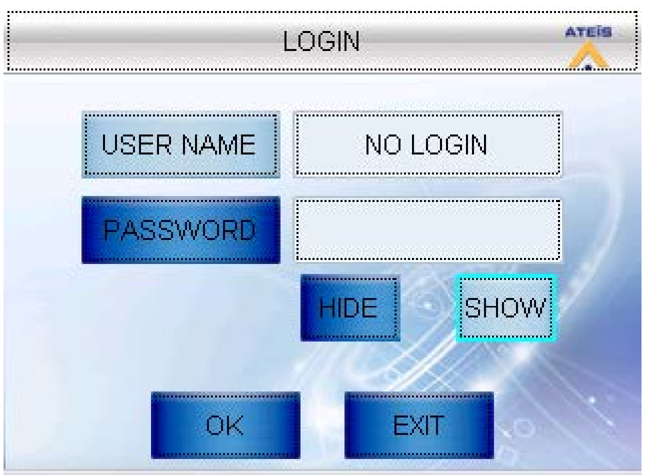
   Click [SYSTEM SET] in [MAIN MENU] page:



Click [LOGIN] in [SYSTEM SET MENU-1] page:



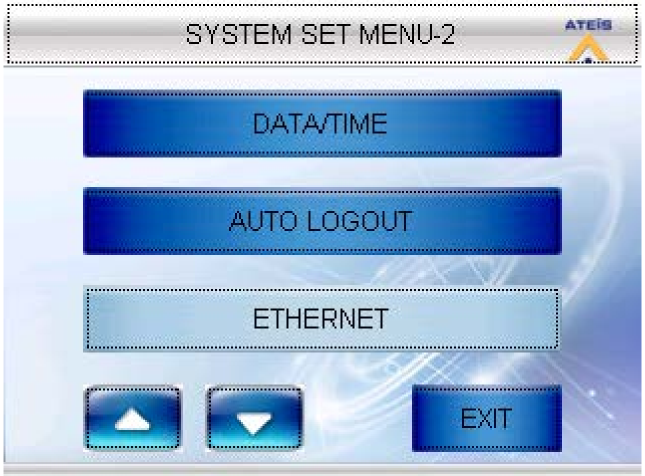
Click [USER NAME] in [LOGIN] page:



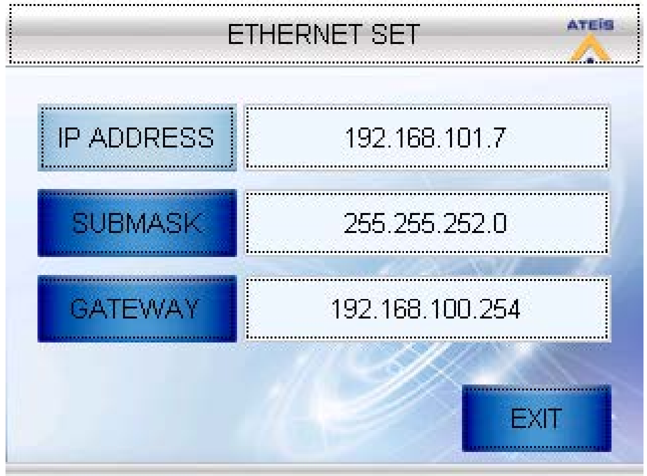
Input ID "ADMIN" which is the default user of IDA8C. Remain the blank of password.

A page showing the message to prompt login is success:

Go back to [SYSTEM SET MENU-2] page, click [ETHERNET]



There are three field: [IP ADDRESS], [SUBMASK] and [GATEWAY]. Click for each to setup.



⚠ You need to reboot device take effect.

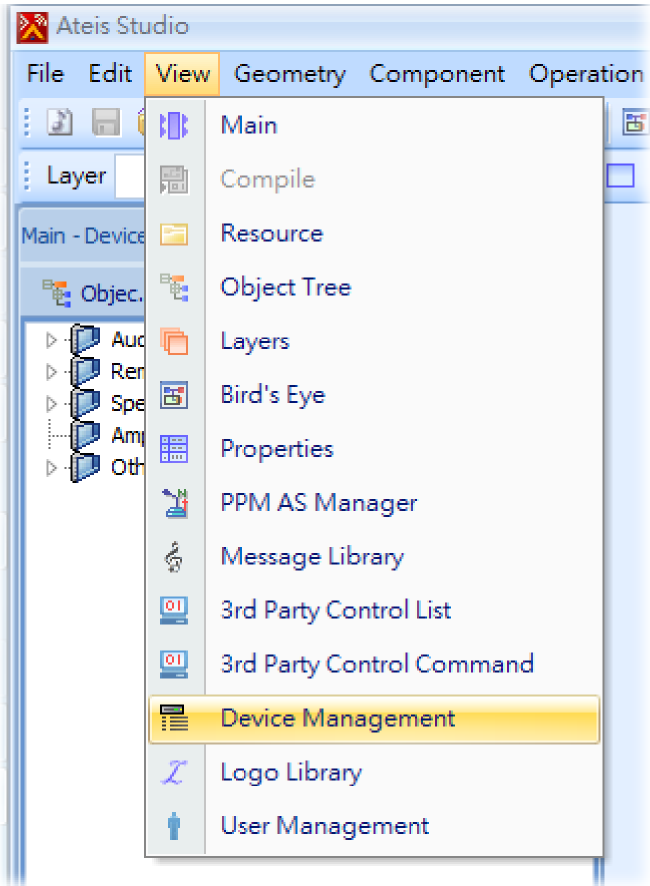
4. Install Ateis Studio

5. Run Ateis Studio

6. Network settings of Ateis Studio

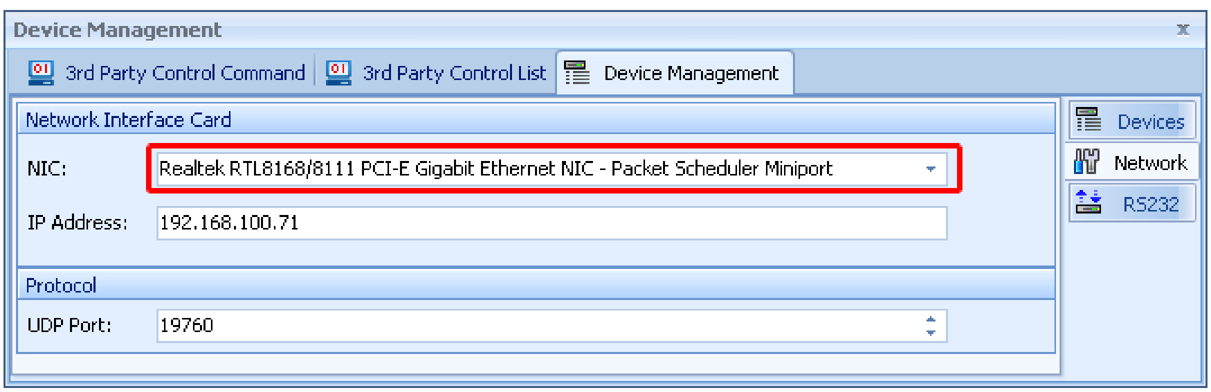
   If there are more than one network interface card on PC, it need to select which one is used to connect to device.

   To open the window for network interface card, click main menu of Ateis Studio [View > Device Management]



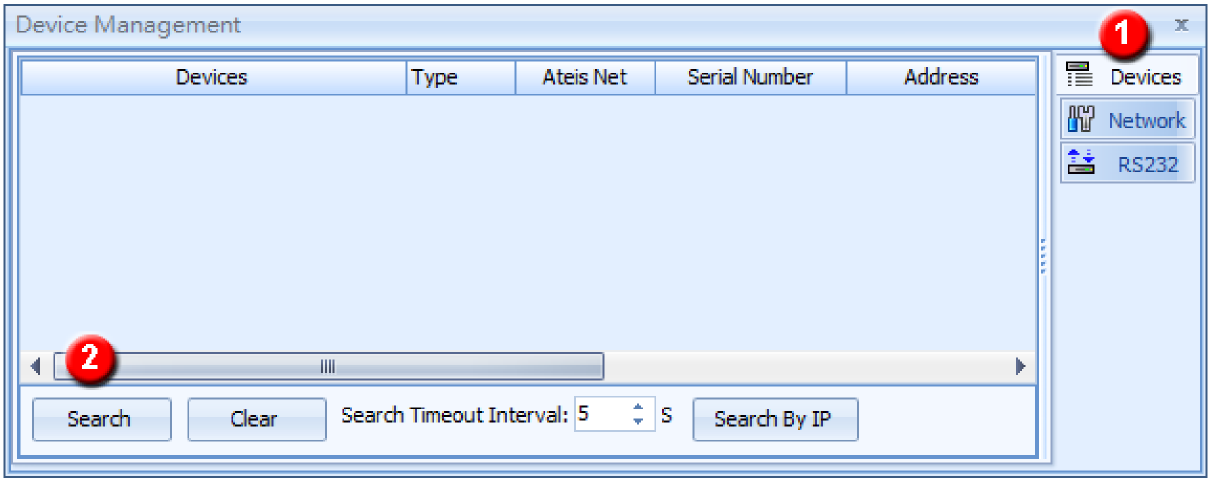
There are three pages on the right side of window, select [Network] page. Choose the right network interface card on NIC field which is labeled using red rectangle.

If UDP port is conflict with the other software, change to other number.

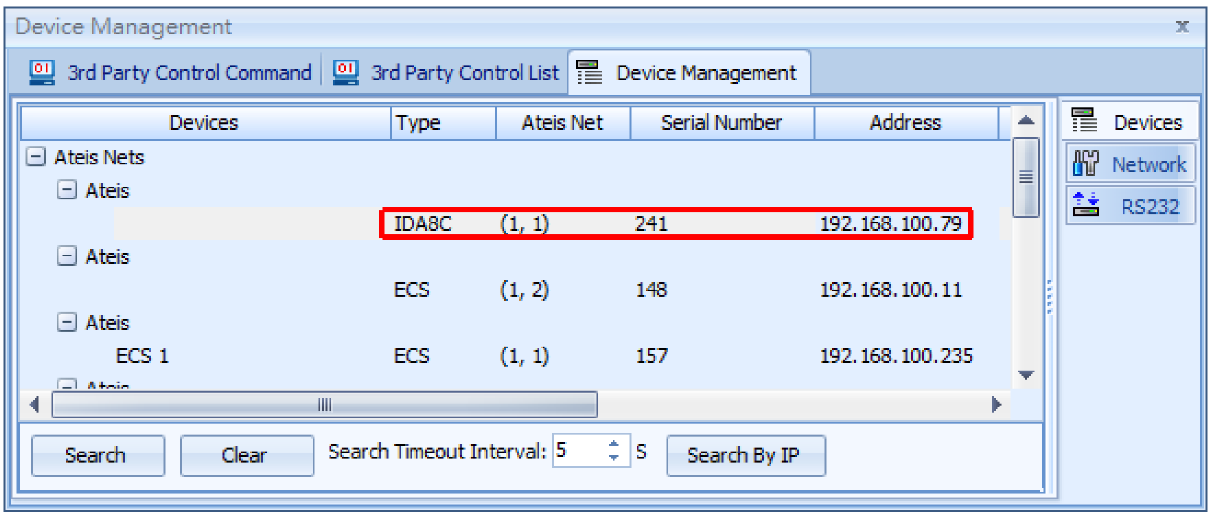


7. Search devices

Go to [Devices] page, and press [Search] button:

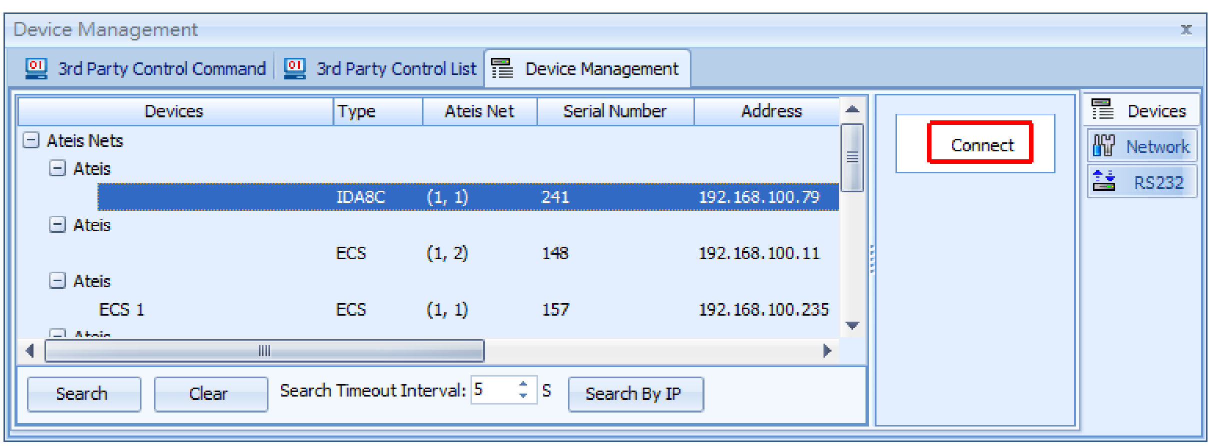


After few seconds, devices are listed by tree structure on [Devices] field, IP address of the device we want to config is 192.168.100.79.

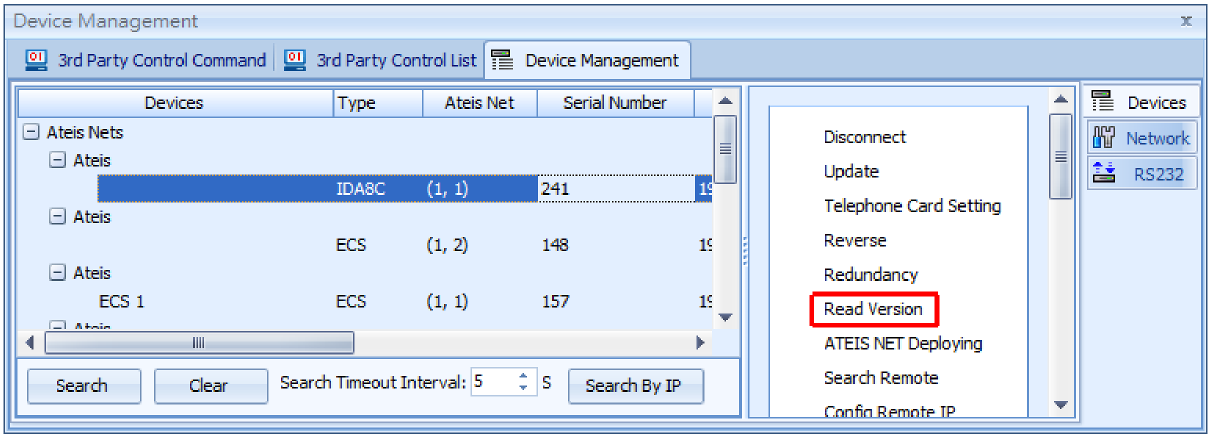


8. Connect to device

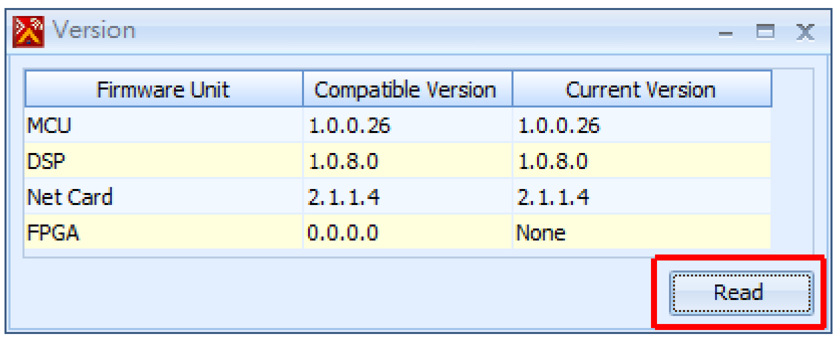
Click [Connect] link to open menu of IDA8C:



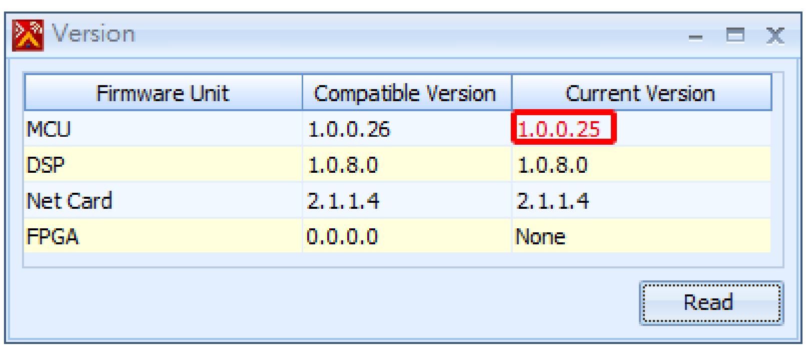
9. Check compatibility between device and Ateis Studio. Click [Read Version] of menu.



A window [Version] opened, press button [Read], And then information about version is display in the grid:

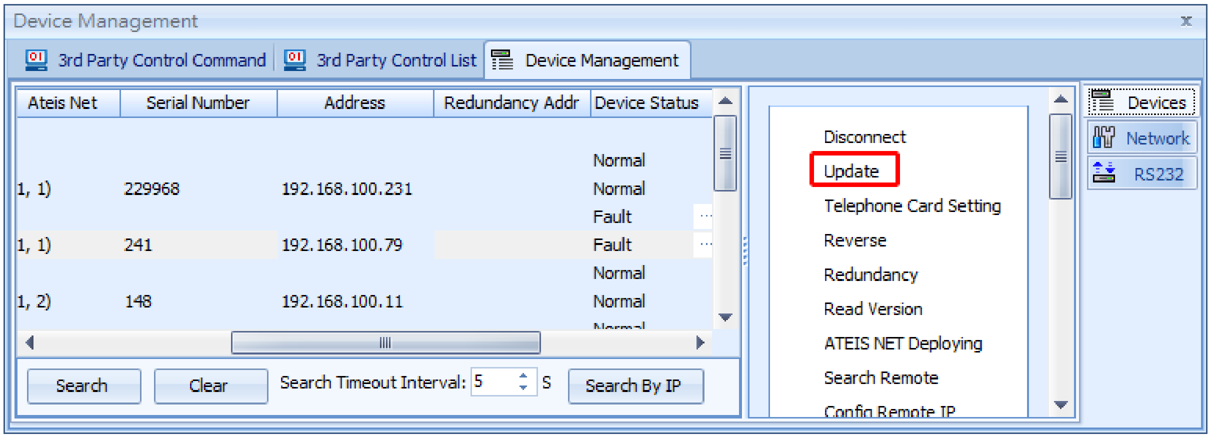


If all firmware unit are compatible with Ateis Studio, go to step 11, otherwise step 10. The below figure is an example of incompatible version of firmware unit See the red rectangle.

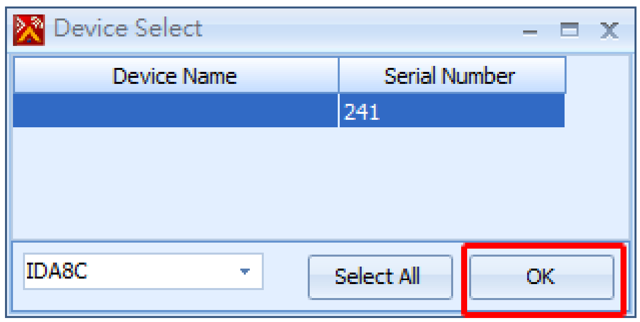


10. Update firmware

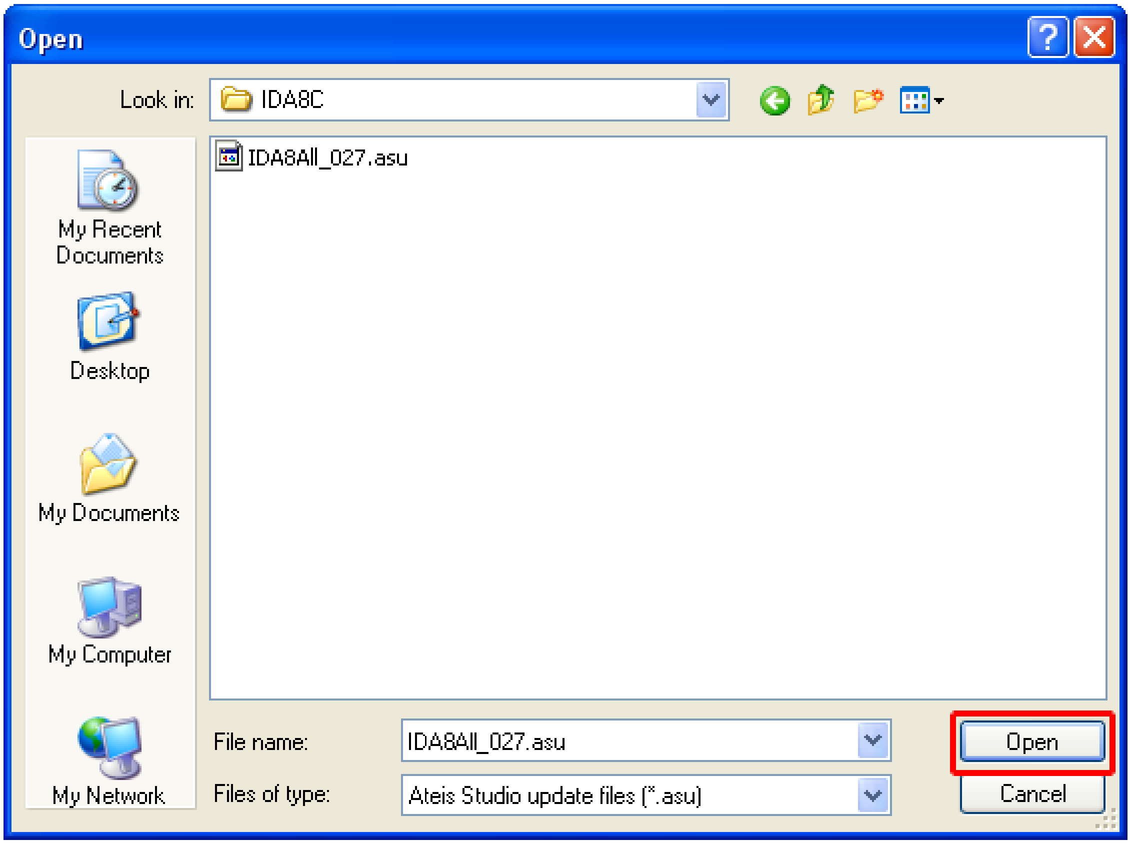
Close the window of [Version], go back to [Device Management] window, click link [Update]:



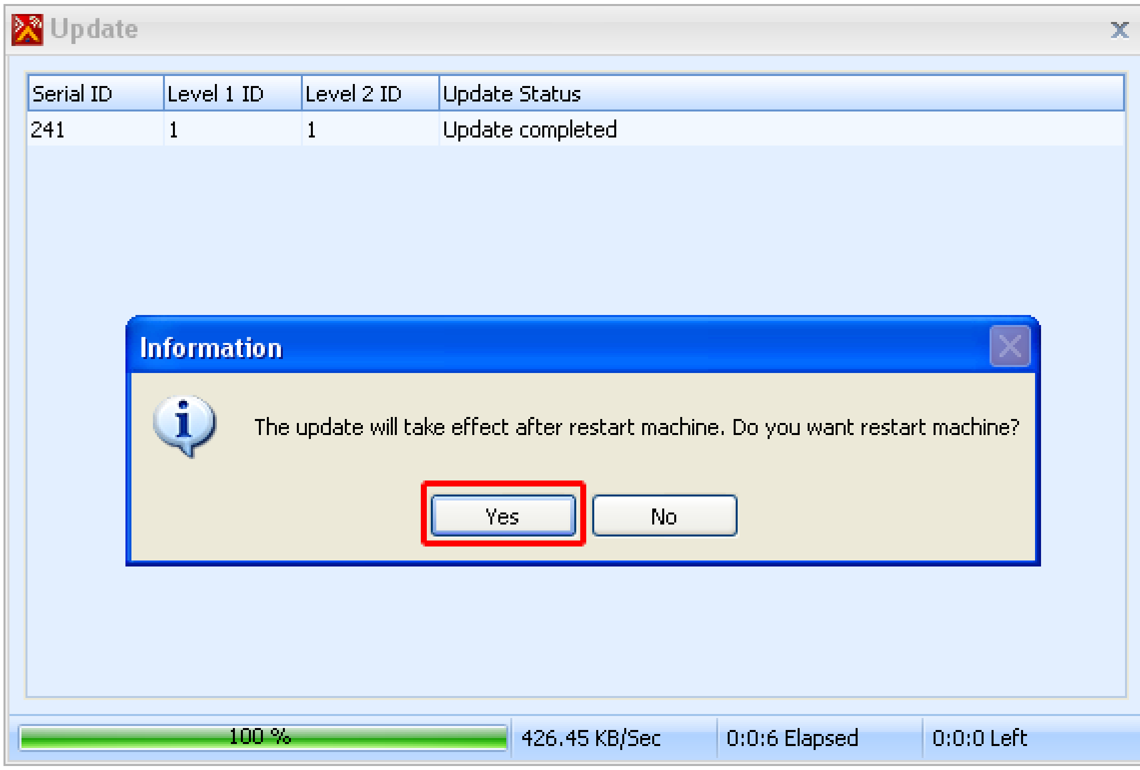
A window [Device Select] opened allow the target IDA8 be selected, click button [OK]:



A file manager opened. Goes into directory [Update file][IDA8C] select .asu file to update.



Then, the update process start, and pop a window after update file is transferred to device, click [Yes] to restart device.



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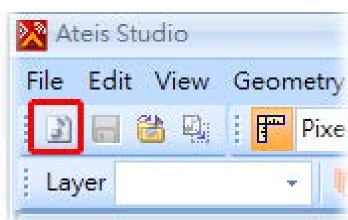
Check device version using step 9 again, the version should become compatible this time, if version is compatible, go to step 11.

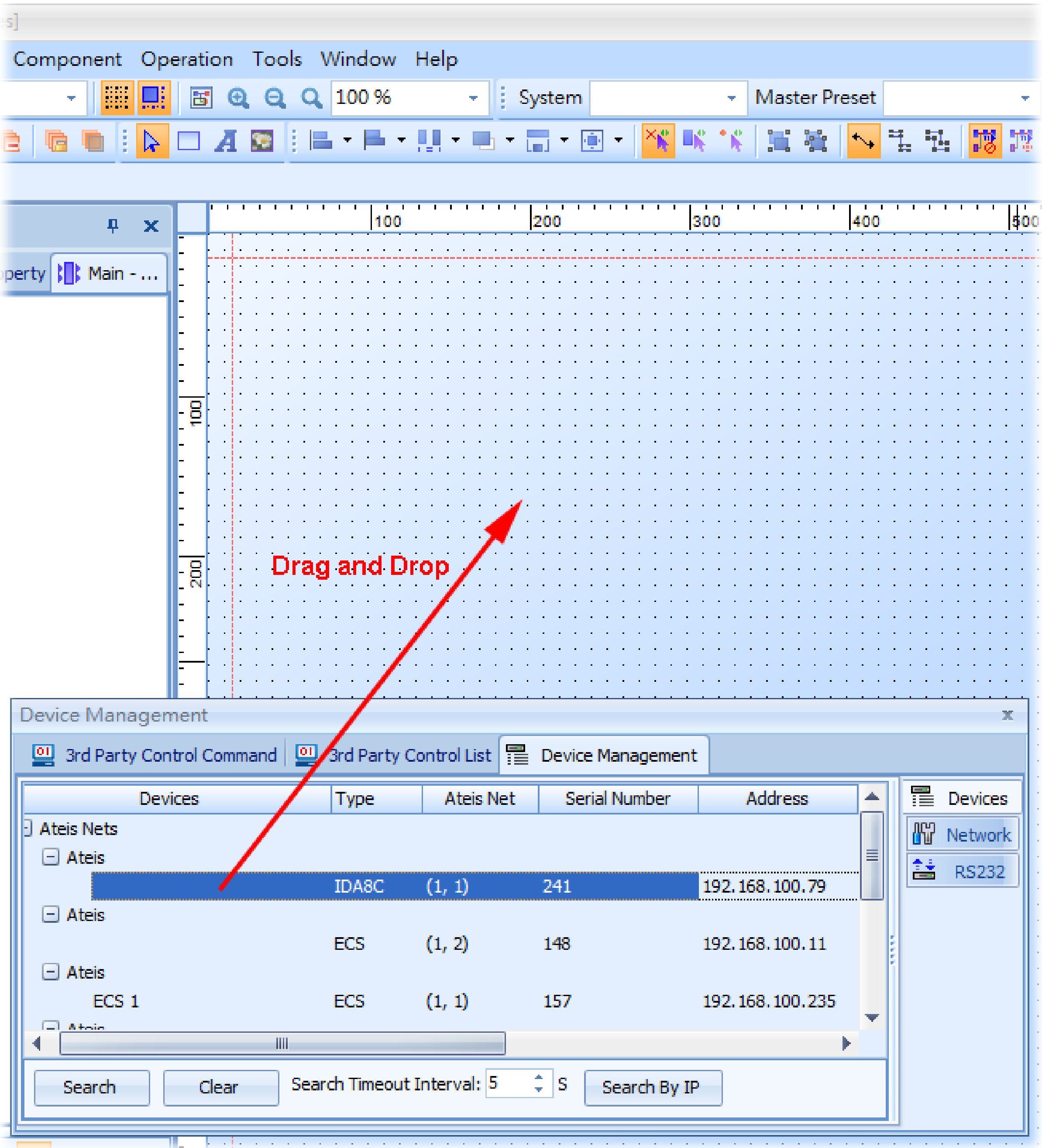
11. Create a New file

Create a new file by clicking the button on left top of Ateis Studio, marked as red rectangle.

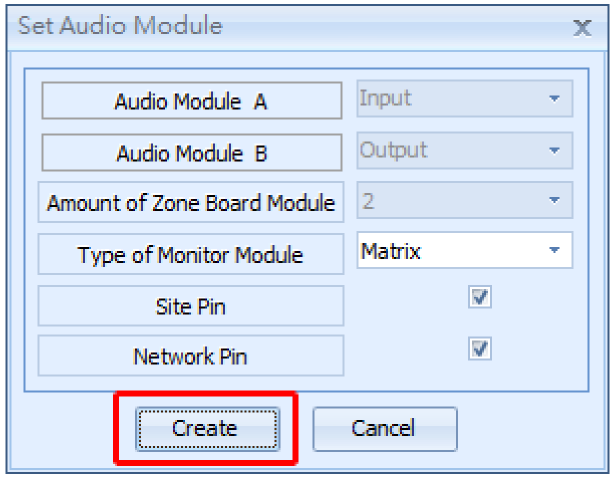


12. Create a IDA8C block in device editor window

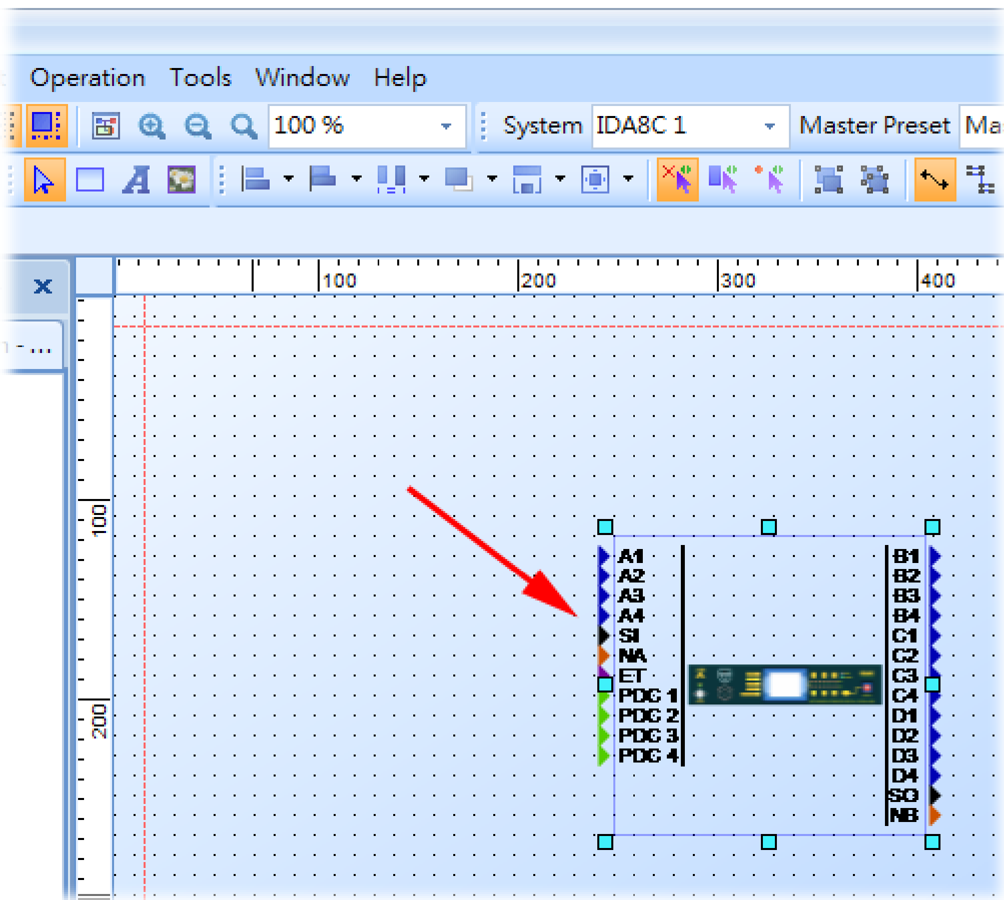
Drag the row in [Device Management] window and drop it in device editor window:



A window [Set Audio Module] opened, click button [Create].

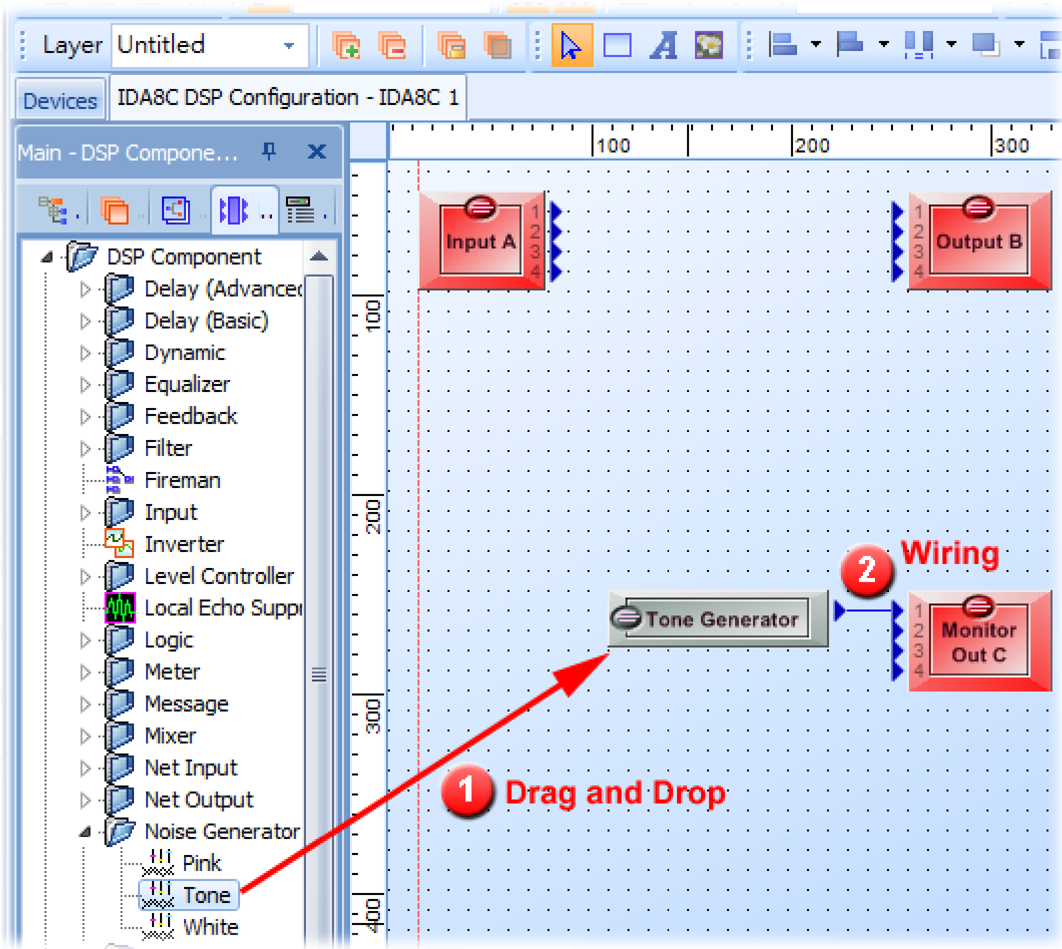


Then a IDA8C block is created for further configuration.



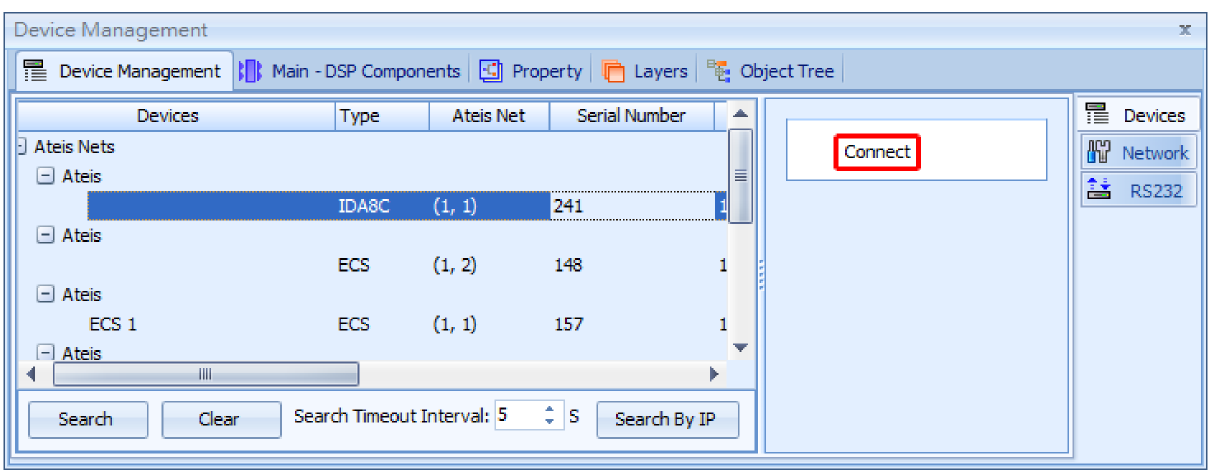
13.Add Tone component and connect to Output component

Double click on IDA8C block to open [IDA8C DSP Configuration - IDA8C1], In this windows, allow you to edit dsp configuration, let's do a simple setup use a Tone Generator and Output component:



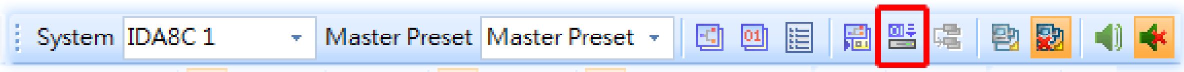
14. Connect

Connect to device.

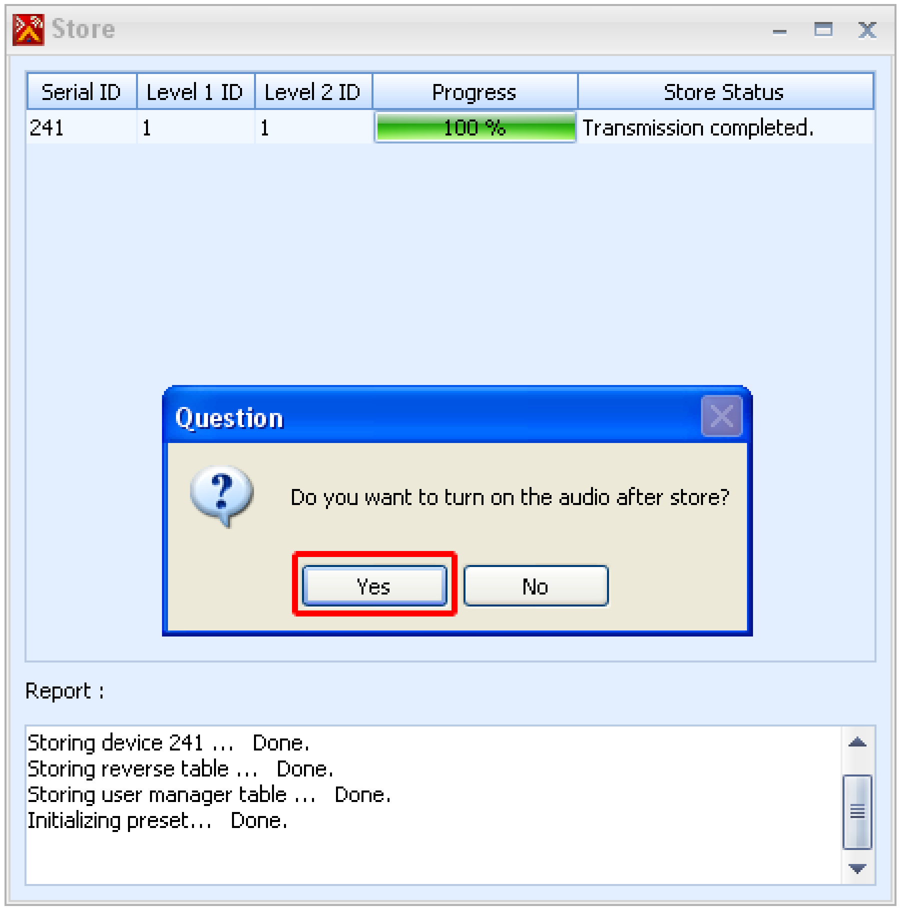


15. Store

Click store button on tools bar.



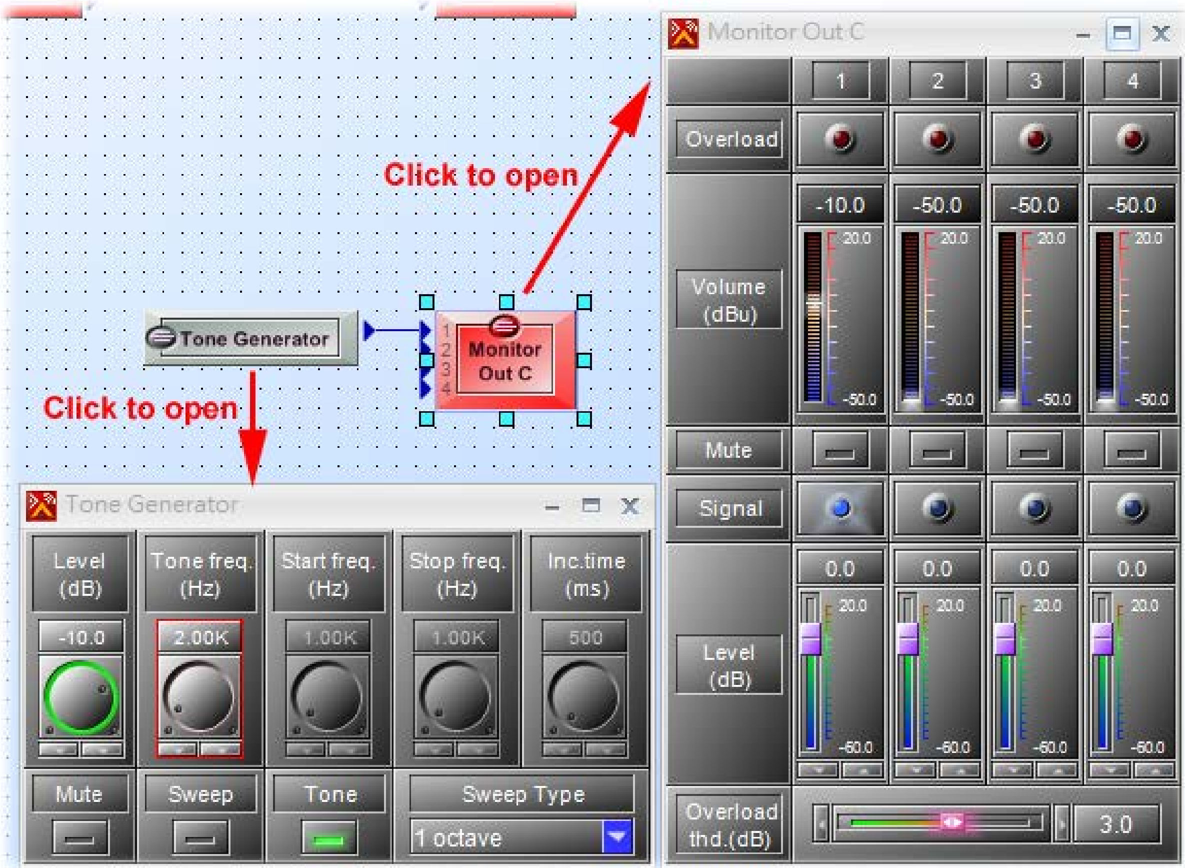
A window [Store] is opened and store process start, during the process, system will ask user to turn on or off audio, click Yes.



16.Adjust element

Set element of Tone Generator:

- Level(dB) = -10
- Tone Button = On
- Tone freq.(Hz) = 2.00k



Now you should see channel 1 meter as figure shown and hear a 2k tone from speaker connect to Amplifier Zone Output(zone1) on rear panel of IDA8C.

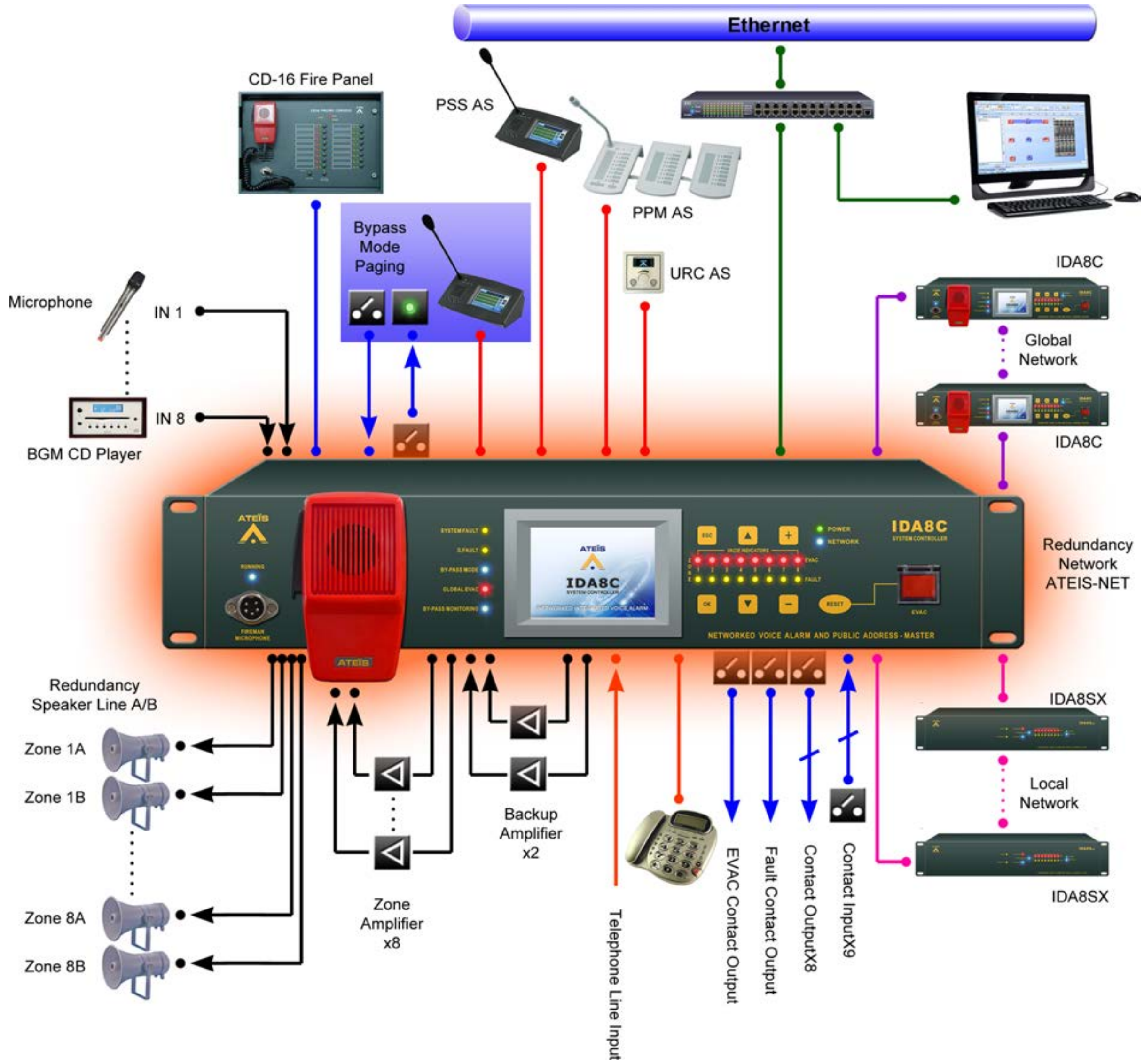
4 Product Features

4.1 IDA8 Series Audio Processor

The IDA8 is a third generation modular system that complies with current architectural demands requiring IP-and/or Fiberoptics Networking to cover for any complex design possible. IDA8 responds to Public Address and Voice Alarm requirements as stated in EN 54-16, ISO 7240-16 and BS 5839/8, with specific attributes for compliance in large installations.

4.1.1 IDA8C

4.1.1.1 Overview



IDA8C Controller unit enables operators to see a detailed overview of the operational status of the entire PA system with a single push button. IDA8C Controller unit is able to run an impedance scan of all components connected to it, covering not only the input paging consoles but including connectors, cabling, processing blocks such as compressors and limiters, delay lines and the network and loudspeakers. It stores a reference measurement of the system as it exists in a given configuration and environment. This reference is subsequently stored in the system. Any alterations to this reference will be reported and are logged in an event log file. User definable thresholds can be applied to these references, allowing for customization to match circumstances.

Being EN 54-16/ISO 7240-16 and BS 5839 part 8 security systems, all components and peripherals are monitored. All incidents are recorded into a data file which can be consulted on the controller module monitor display or on a PC using the ATEÏS Studio remote control software. Also, any detected fault is signalled by a general fault output contact available on the IDA8C Controller unit. A built-in loudspeaker output enables selective listening to all the sources and system's 100 V output signals.

IDA8C Controller unit is easily configured with our PC based ATEÏS-Studio global software (Windows compatible). Software access can be password protected. Once programmed, the system will be able to work independently(off-line) without the need of a PC to be connected.

4.1.1.2 Front Panel

1 Fireman Microphone Active LED

Show the activity of the fireman microphone, there are two states of this LED: test

Status	Frequency	Activity
Permanent		Active, Fireman microphone is allowed to paging.
Blinking		Zones are busy, fireman microphone is not allowed to paging.
Blinking		Pre chime is playing.

2 Fireman Microphone Input

DIN Connector for fireman microphone connection.

3 Fireman Microphone Hook

An U-shape hook to place fireman microphone.

4 Monitoring Speaker

The IDA8C integrates a monitoring speaker in front panel. There are two functions of this speaker

Function	Activity
Warning Tone	A buzzer sound is generated when some faults is detected in system.
Source Monitoring	Monitor audio sources of IDA8C. The source to be monitored can be selected via Ateis Studio software.

Note: If a fault is detected when speaker is monitoring source, then the source monitoring will be interrupted and play buzzer sound for the fault.

5 Bypass Mode LED

To indicate IDA8C is in bypass mode or not.

6 G.Fault LED

This LED lights on when a global fault is detected.

Note: Fault LED and system fault LED are exclusive, if system fault and global fault are detected at the same time, system only light the system fault LED.

7 System Fault LED

This LED lights on when a system fault fault is detected.

8 Global EVAC LED

To indicate system is in EVAC paging or not. This LED lights on if any IDA8 device over Ateis Net do an EVAC paging.

9 Bypass Monitoring LED

Show the monitoring function is enabled or not. This LED lights on if monitoring of IDA8 is disabled by user.

10 Touch Screen

3.5" Color touch panel offer a graphical interface for user to control or read the status of system. There is a multi layered menu on it to get/set parameters or change settings of device.

11 ESC Button

To Navigate menu of touch screen.

12 OK Button

To Navigate menu of touch screen.

13 Up Button

To Navigate menu of touch screen.

14 Down Button

To Navigate menu of touch screen.

15 Plus Button

To Navigate menu of touch screen.

16 Down Button

To Navigate menu of touch screen.

17 Zone EVAC LEDs

To show the audio channel is in EVAC paging or not. Each LED is correspond to a output channel of Network Paging component, for example, 1st LED is for pin M1, 2nd LED for pin M2 and so on.

18 Zone Fault LEDs

To show fault status of zone. This LED light on if one of following faults is detected:

- Normal AMP Error
- Line A Error
- Line B Error
- Backup AMP Error
- AMP Line Leakage Error

19 EVAC Reset Button

Cancel the event triggered by EVAC button.

20 Power LED

This LED lights on when IDA8C is power on.

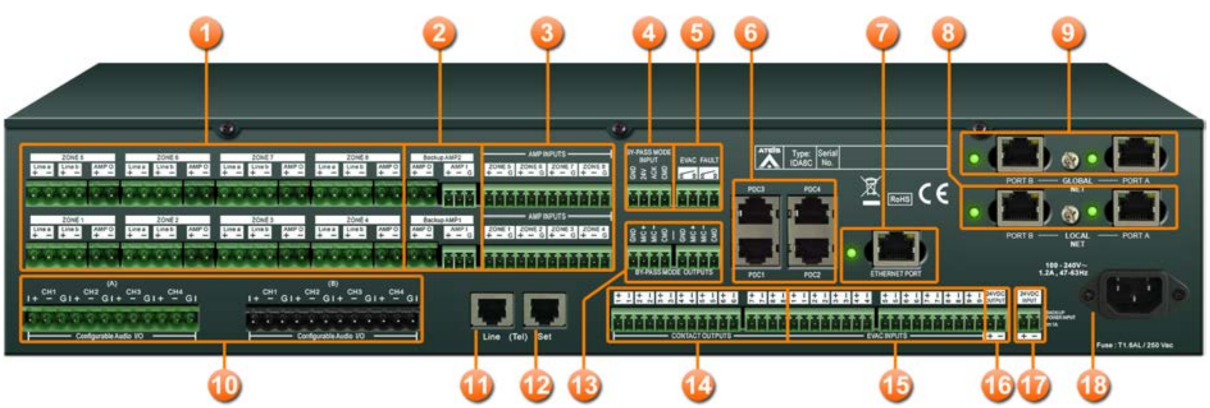
21 Network LED

Show the status of Ateis Net. This LED lights on if more than two deployed IDA8 devices in Ateis Net.

22 EVAC Button

To trigger event bind with this button.

4.1.1.3 Rear Panel



1 Speaker Zone Outputs

There are 8 zones for speaker & amplifier connection. each zone consists of following connectors (from left to right):

- connector of line A speaker
- connector of line B speaker

- connector of 100V audio coming from amplifier

IDA8 output exactly the same audio signal to line A and B. Two speakers can connect to one zone for redundancy purpose, one connect to line A, another connect to line B. Both speaker lines are monitored, that's mean if a damage of speaker is detected, a global fault "Line A/B Error" will be generated.

2 Backup Amplifier I/Os

Two backup amplifier connectors included 0 dB to amplifiers and 100V return from amplifiers.

3 Amplifier Zone Outputs

8 Zone 0dB output to amplifiers.

4 Bypass Mode Input

- Contact input to engage the bypass mode paging(CMD).
- Contact output to display the state of bypass mode paging(ACK).

5 EVAC, Fault State Outputs

- EVAC Output Contact: this contact is closed if system is under EVAC paging.
- Fault Output Contact: this contact is open if a fault is detected.

6 PDC(Peripherals Device Controller) Connectors

Four RJ-45 connectors to connect consoles or peripheral devices. For examples, PSS AS, URC AS, PPM AS, ... are connect to IDA8C via this connector.

7 Ethernet Connector

RJ-45 connector to link IDA8C on an ethernet network. The following end points are communicated with IDA8C through ethernet networking.

- Ateis Studio software
- 3rd party devices
- Voxnet Server
- PMIP-D
- VNB
- PPM-IT5
- TERRA Devics
- Modbus Protocol

8 Local Ateis Net Connectors

Optional card to build a local dedicated IDA8C-IDA8S network.

9 Global Ateis Net Connectors

Optional card to build a global dedicated IDA8C network.

10 Configurable Audio I/Os

Two configurable 0dB audio I/O port A and B. Each port is available to assemble an audio card. There are 4 channels on each audio I/O card.

11 Telephone Line Input

A connector for telephone signal coming from telephone company.(optional).

12 Telephone Connector

A connector for external telephone connection.(optional).

13 Bypass Mode Outputs

This port to share the bypass mode microphone signals through the IDA8C-IDA8S network(only needed when using fiber optic network).

14 Contact Outputs

8 logic outputs channels to close/open circuit for an external device, this contact is normally open.

15 Evacuation Inputs

9 evacuation contact inputs that allow the monitoring of external contact. They also can be used in UGA mode, trigger by a voltage polarization change.

16 24V DC Output

This connector supplies a 24VDC source.

17 24V DC Input

Main 24VDC power supply connector.

18 AC Power Socket

Main 110~240 V 1.2A, 47~63Hz AC power input with fuse. If 24V DC and AC power input at the same time, IDA8C will use AC power, and switch to DC power if there is no AC power input.

Fuse Rating: 3.15A

❖ Zone Boards

- Zone Board1

  The following I/Os are belong to zone board1 in IDA8C: speaker zone output ch1~ch4, backup amplifier1, amplifier zone output ch1 ~ ch4.

- Zone Board2

  The following I/Os are belong to zone board2 in IDA8C: speaker zone output ch5~ch8, backup amplifier2, amplifier zone output ch5 ~ ch8.

4.1.1.4 Characteristics

❖Case

- Dimension = 436mm (W) x 313mm (L) x 88mm (H).
- Weight = 5.36Kg.
- Color = RAL7016.

❖Screen

- 3.5" Color touch panel.

❖Power

Item	Voltage	Current Consumption	Comment
AC Input	100V~240V	1.6A@100V, 0.7A@240V	Frequency:47Hz~63Hz
DC Input	18V~26V, Typical 24V	2A	-
DC Output	18V~26V, Typical 24V	0.5A	-

- AC Maximum Consumption = 45 W.

❖Amplifier Zone Outputs

- Maximum level = 14 dB
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Speaker line/Amplifier

- Maximum Power = 700W(RMS).
- Maximum Power(Siren + Message) = 1000W.

❖Monitoring Speaker

- Impedance = 8 Ohm.
- Maximum Power @ 1kHz = 1.6W.

- THD @ 1kHz = < 1%.
- Bandwidth @ -3dB = 100Hz ~ 12kHz.

❖ Configurable Audio Output

- Maximum level = 14 dBu
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

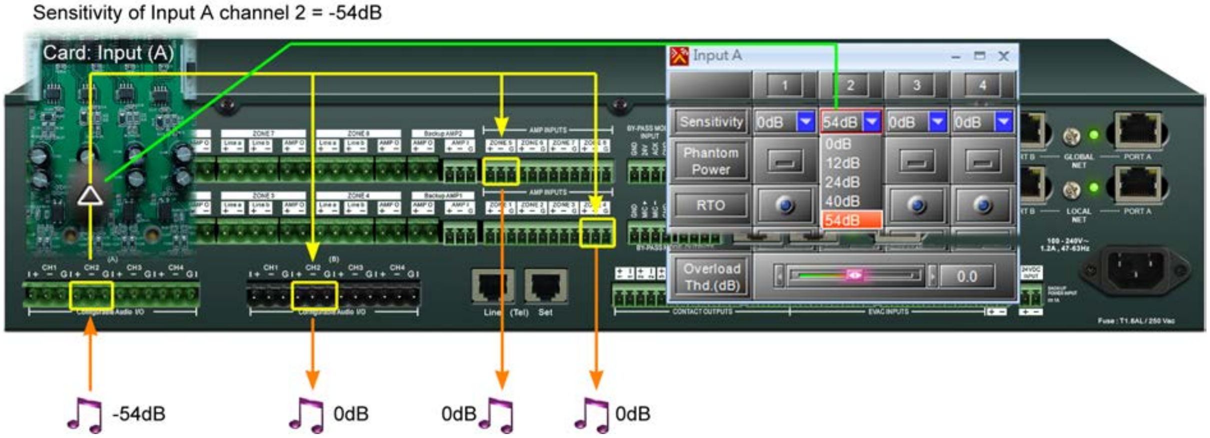
Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖ Configurable Audio Input

- Sensitivity

For each channel of configurable audio input card, there are five levels to gain the audio signal. They are sensitivity values of 0dBu, -12dbBu, -24dBu, -40dBu and -54dBu respectively. The meaning of sensitivity value is how large the gain for an audio source, and to amplify the giving minus input source to 0dBu, the figure below is an example when the sensitivity value is set to -54dBu, and input source is -54dBu, after the gaining circuit of audio card, you'll get the 0dBu of output.

Sensitivity of Input A channel 2 = -54dB



- Maximum level = 14 dBu

- Input Impedance = 10k Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.

❖PDC

- Maximum Output Level = 10 dBu.
- Output Impedance = 300 Ohm.
- THD @ 1kHz at output < 0.02.
- Bandwidth @ -3dB at output = 20Hz ~ 20kHz.
- Noise @ 22Hz ~ 22kHz = -85dBu.
- Maximum Input Level = 16 dBu.
- Input Impedance = 11k Ohm.
- THD @ 1kHz at input < 0.02
- Bandwidth @ -3dB at input < 20Hz ~ 22kHz.
- RS232 voltages = -6.5/+6.5

❖Evacuation Inputs(Contact Mode)

- Bias voltage:

Item	Minimum	Maximum	Unit
Voltage	-	5	VDC

- Monitoring resistor = 4.7k Ohm.

❖Evacuation Inputs(Voltage Mode)

On Voltage:

Item	Minimum	Maximum	Unit
Voltage	18	72	VDC

❖Contact Outputs + EVAC, Fault State Outputs

Item	Minimum	Maximum	Unit
Voltage	-	100	VDC
Current	-	0.5	ADC

❖Working Temperature.

0°C ~ 40°C

4.1.1.5 Peripherals

Following table is the peripherals supported by IDA8C:

Device	Connection	Max. Num	Function
Fireman MIC	Fireman MIC Input	1	Paging where operator is close to IDA8C.
PPM AS	PDC	31/PDC	Remote console for with paging ability.
PSS AS	PDC	1/PDC	Remote console for Paging, Event Triggering(Element Control, Master/Sub Preset Control).
URC AS	PDC	1/PDC	Remote console for Master/Sub Preset Controlling, Element Controlling.
URGP	PDC	1/PDC	Evacuation Input extension.
DNM	PDC	1/PDC	Auto noise gain for audio signals.
PPM-IT5	Ethernet	1 Active Over Eth	Paging, Element Control, Master/Sub Preset Control.
URC200 TPC	Ethernet	Eth. Limit	Parameters Control, Master/Sub Preset Control.
CD8	PDC	31/PDC	Wall mounted cabinet remote paging console with 8 buttons/ zones.
CD16	PDC	31/PDC	Wall mounted cabinet remote paging console with 16 buttons/ zones.
CD Touch	PDC	1/PDC	Wall mounted cabinet remote paging console with touch screen and Fireman Microphone.
PCP	PDC	31/PDC	Wall mounted cabinet remote paging console with telephone styled microphone.
CDPA	PDC	31/PDC	Wall mounted cabinet remote paging console with 24 buttons/ zones and 2 extra selectable microphone sources.
PSC	PDC	31/PDC	A grouped console is comprised of a pad with monitoring speaker, a pad with a gooseneck and 8 buttons/zones, two pads with 8 buttons/zones.
Deskpad	PDC	1/PDC	A remote dialer for making telephone calls via IDA8's telephone hardware.
Deskpad Box	PDC	1/PDC	It's a RF transceiver to communication between Deskpad and IDA8, use it, Deskpad can working without physcial connection.

Notes:

Eth. Limit: The maximum numbers of IP that can be assigned over the network.

4.1.1.6 3rd Party Control

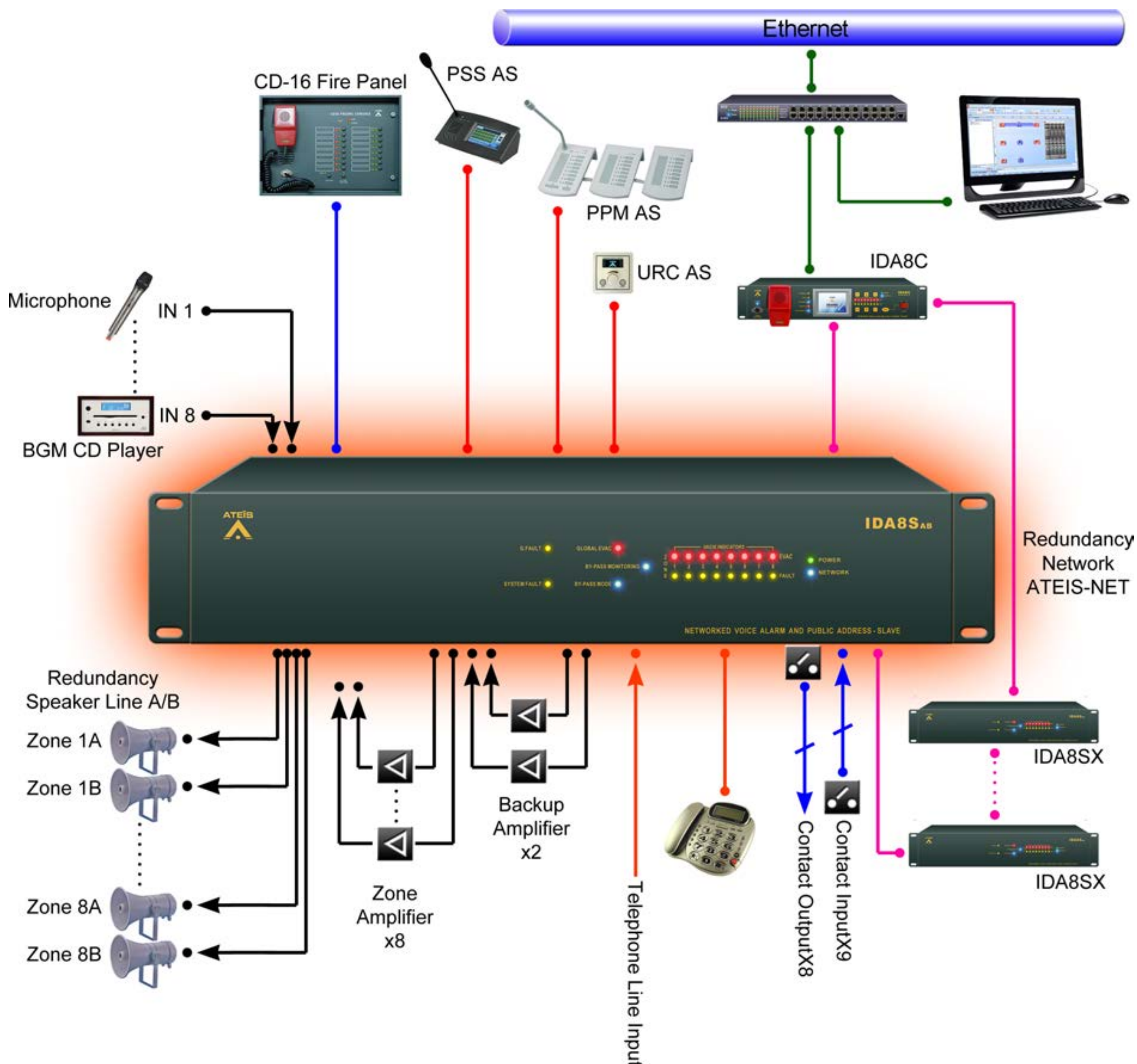
The following table list 3rd Party Control protocols IDA8C Supported.

Protocol	Interface		Function
	Connection	Settings	
Ateis 3rd Party Protocol	Ethernet(UDP)	UDP Port = 19761	• Read/Write value of parameters. • Stepped adjustment for parameters.
Modbus	Ethernet(TCP) PDC(RS485)	TCP Port = 502 PDC Port 1~ 4	• Level control • Paging

Protocol	Interface		Function
	Connection	Settings	
			• Read Monitoring Status • Read Evacuation Status • Play Message • Logic I/O Control

4.1.2 IDA8SAB

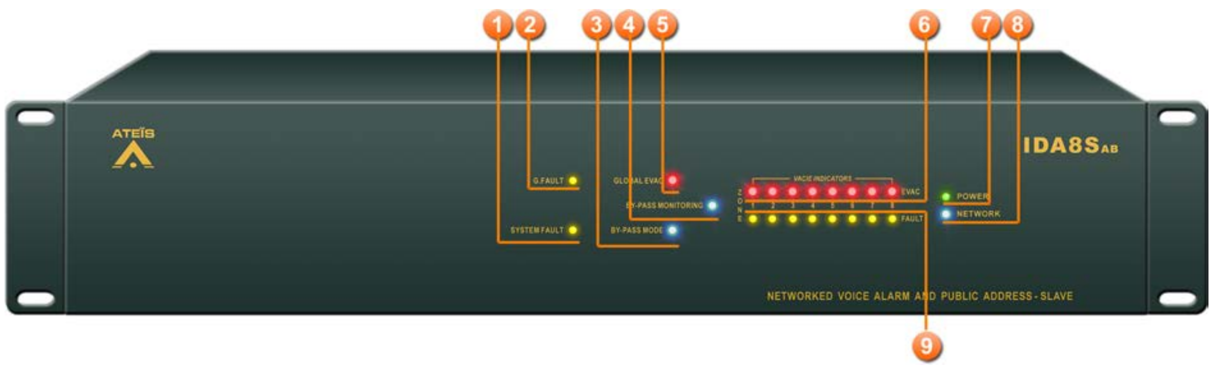
4.1.2.1 Overview



The IDA8SAB Slave units provide audio in-and output expansion of the IDA8 Systems using a secured 48-channel audio and data network over CAT5 or fiber optic. Each IDA8SAB expands the IDA8 System with an additional 8 audio inputs and outputs, 2 additional security microphone consoles PSS AS and security programmable switching contacts.

The Network cards that comes with the unit provides a redundant 48 channels audio and data connection, Ateïs- Net, between one controller and a maximum of 32 IDA8SX Slave units in one rack system.

4.1.2.2 Front Panel



1 System Fault LED

This LED lights on when a system fault fault is detected.

2 G.Fault LED

This LED lights on when a global fault is detected.

3 Bypass Mode LED

To indicate IDA8C is in bypass mode or not.

4 Bypass Monitoring LED

Show the monitoring function is enabled or not. This LED lights on if monitoring of IDA8 is disabled by user.

5 Global EVAC LED

To indicate system is in EVAC paging state or not. This LED lights on if any IDA8 device over Ateis Net do an EVAC paging. See the topic of Network Paging Component to learn more about how to config a EVAC paging.

6 Zone EVAC LEDs

To show the audio channel is in EVAC paging or not. Each LED is correspond to a output channel of Network Paging component, for example, 1st LED is for pin M1, 2nd LED for pin M2 and so on.

7 Power LED

This LED lights on when IDA8SAB is power on.

8 Network LED

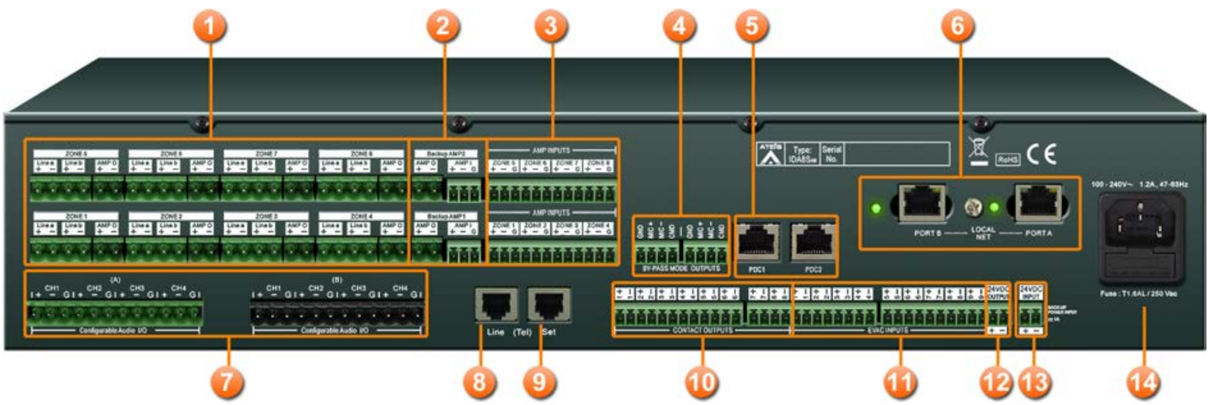
Show the status of Ateis Net. This LED lights on if more than two deployed IDA8 devices in Ateis Net.

9 Zone Fault LEDs

To show fault status of zone. This LED light on if one of following faults is detected:

- Normal AMP Error
- Line A Error
- Line B Error
- Backup AMP Error
- AMP Line Leakage Error

4.1.2.3 Rear Panel



1 Speaker Zone Outputs

There are 8 zones for speaker & amplifier connection. each zone consists of following connectors (from left to right):

- connector of line A speaker.
- connector of line B speaker.
- connector of 100V audio coming from amplifier.

IDA8 output exactly the same audio signal to line A and B. Two speakers can connect to one zone for redundancy purpose, one connect to line A, another connect to line B. Both speaker lines are monitored, that's mean if a damage of speaker is detected, a global fault "Line A/B Error" will be generated.

2 Backup Amplifier I/Os

Two backup amplifier connectors included 0 dB to amplifiers and 100V return from amplifiers.

3 Amplifier Zone Outputs

8 Zone 0dB output to amplifiers.

4 Bypass Mode Outputs

This port to share the bypass mode microphone signals through the IDA8C-IDA8S network(only needed when using fiber optic network).

5 PDC(Peripherals Device Controller) Connectors

Two RJ-45 connectors to connect consoles or peripheral devices. For examples, PSS AS, URC AS, PPM AS, ... are connect to IDA8SAB via this connector.

6 Local Ateis Net Connectors

Optional card to build a local dedicated IDA8C-IDA8S network.

7 Configurable Audio I/Os

Two configurable 0dB audio I/O port A and B. Each port is available to assemble an audio card. There are 4 channels on each audio I/O card.

8 Telephone Line Input

A connector for telephone signal coming from telephone company.(optional).

9 Telephone Connector

A connector for external telephone connection.(optional).

10 Contact Outputs

8 logic outputs channels to close/open circuit for an external device, this contact is normally open.

11 Evacuation Inputs

9 evacuation contact inputs that allow the monitoring of external contact. They also can be used in UGA mode, trigger by a voltage polarization change.

12 24V DC Output

This connector supplies a 24VDC source.

13 24V DC Input

Main 24VDC power supply connector.

14 AC Power Socket

Main 110~240 V 1.2A, 47~63Hz AC power input with fuse. If 24V DC and AC power input at the same time, IDA8C will use AC power, and switch to DC power if there is no AC power input.

Fuse Rating: 1.6A

4.1.2.4 Characteristics

❖ Case

- Dimension = 436mm (W) x 289mm (L) x 88mm (H).
- Weight = 5Kg.
- Color = RAL7016.

❖ Power

Item	Voltage	Current Consumption	Comment
AC Input	100V~240V	1.2A@100V, 0.5A@240V	Frequency:47Hz~63Hz

DC Input	18V~26V, Typical 24V	2A	-
DC Output	18V~26V, Typical 24V	0.5A	-

- AC Maximum Consumption = 45 W.

❖Amplifier Zone Outputs

- Maximum level = 14 dB
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Speaker line/Amplifier

- Maximum Power = 700W(RMS).
- Maximum Power(Siren + Message) = 1000W.

❖Configurable Audio Output

- Maximum level = 14 dBu
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

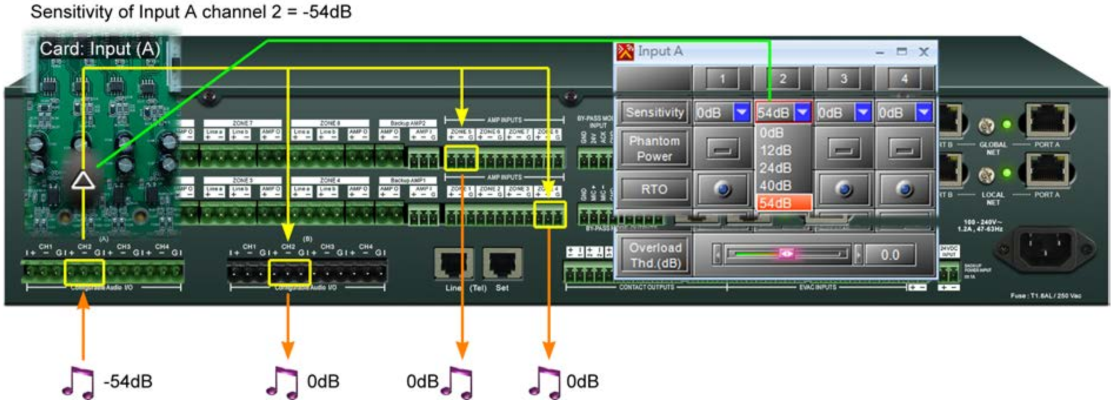
Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Configurable Audio Input

- Sensitivity

For each channel of configurable audio input card, there are five levels to gain the audio signal. They are sensitivity values of 0dBu, -12dbBu, -24dBu, -40dBu and -54dBu respectively. The meaning of sensitivity value is how large the gain for an audio source, and to amplify the giving minus input source to 0dBu, the figure below is an example when the sensitivity value is set to -54dBu, and input source is -54dBu, after the gaining circuit of audio card, you'll get the 0dBu of output.

Sensitivity of Input A channel 2 = -54dB



- Maximum level = 14 dBu
- Input Impedance = 10k Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.

❖PDC

- Maximum Output Level = 10 dBu.
- Output Impedance = 300 Ohm.
- THD @ 1kHz at output < 0.02.
- Bandwidth @ -3dB at output = 20Hz ~ 20kHz.
- Noise @ 22Hz ~ 22kHz = -85dBu.
- Maximum Input Level = 16 dBu.
- Input Impedance = 11k Ohm.
- THD @ 1kHz at input < 0.02
- Bandwidth @ -3dB at input < 20Hz ~ 22kHz.
- RS232 voltages = -6.5/+6.5

❖Evacuation Inputs(Contact Mode)

- Bias voltage:

Item	Minimum	Maximum	Unit
Voltage	-	5	VDC

- Monitoring resistor = 4.7k Ohm.

❖Evacuation Inputs(Voltage Mode)

On Voltage:

Item	Minimum	Maximum	Unit
Voltage	18	72	VDC

❖Contact Outputs + EVAC, Fault State Outputs

Item	Minimum	Maximum	Unit
Voltage	-	100	VDC
Current	-	0.5	ADC

❖Working Temperature.

- 0°C ~ 40°C

4.1.2.5 Peripherals

Following table is the peripherals supported by IDA8SAB:

Device	Connection	Max. Num	Function
Fireman MIC	Fireman MIC Input	1	Paging where operator is close to IDA8C.
PPM AS	PDC	31/PDC	Remote console for with paging ability.
PSS AS	PDC	1/PDC	Remote console for Paging, Event Triggering(Element Control, Master/Sub Preset Control).
URC AS	PDC	1/PDC	Remote console for Master/Sub Preset Controlling, Element Controlling.
URGP	PDC	1/PDC	Evacuation Input extension.

Device	Connection	Max. Num	Function
DNM	PDC	1/PDC	Auto noise gain for audio signals.
PPM-IT5	Ethernet	1 Active Over Eth	Paging, Element Control, Master/Sub Preset Control.
URC200 TPC	Ethernet	Eth. Limit	Parameters Control, Master/Sub Preset Control.
CD8	PDC	31/PDC	Wall mounted cabinet remote paging console with 8 buttons/ zones.
CD16	PDC	31/PDC	Wall mounted cabinet remote paging console with 16 buttons/ zones.
CD Touch	PDC	1/PDC	Wall mounted cabinet remote paging console with touch screen and Fireman Microphone.
PCP	PDC	31/PDC	Wall mounted cabinet remote paging console with telephone styled microphone.
CDPA	PDC	31/PDC	Wall mounted cabinet remote paging console with 24 buttons/ zones and 2 extra selectable microphone sources.
PSC	PDC	31/PDC	A grouped console is comprised of a pad with monitoring speaker, a pad with a gooseneck and 8 buttons/zones, two pads with 8 buttons/zones.
Deskpad	PDC	1/PDC	A remote dialer for making telephone calls via IDA8's telephone hardware.
Deskpad Box	PDC	1/PDC	It's a RF transceiver to communication between Deskpad and IDA8, use it, Deskpad can working without physcial connection.

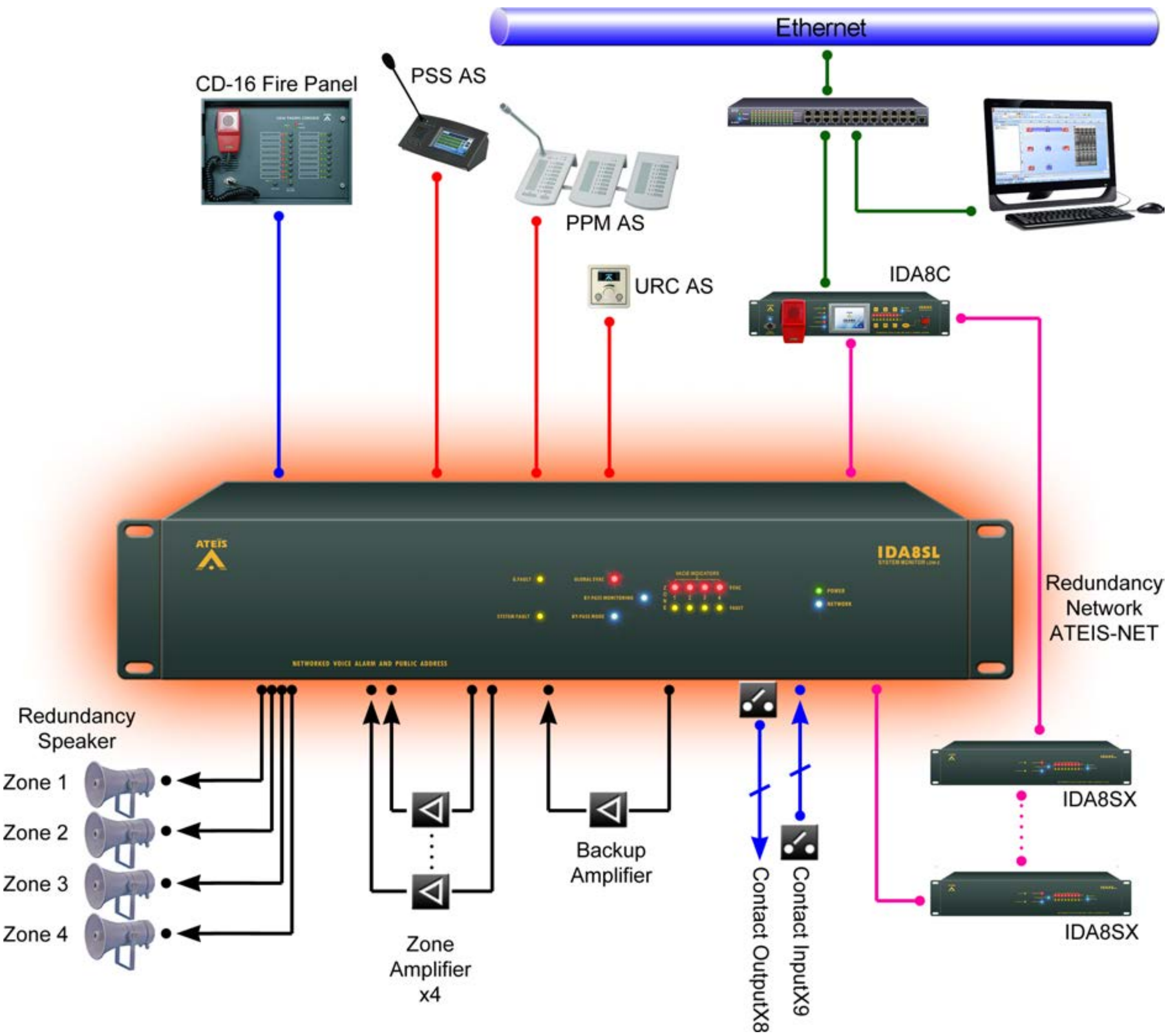
4.1.2.6 3rd Party Control

The following table list 3rd Party Control protocols IDA8C Supported.

Protocol	Interface		Function
	Connection	Settings	
Ateis 3rd Party Protocol	Ethernet(UDP)	UDP Port = 19761	• Read/Write value of parameters. • Stepped adjustment for parameters.
Modbus	PDC(RS485)	PDC Port 1~ 2	• Level control • Paging • Read Monitoring Status • Read Evacuation Status • Play Message • Logic I/O Control

4.1.3 IDA8SL

4.1.3.1 Overview

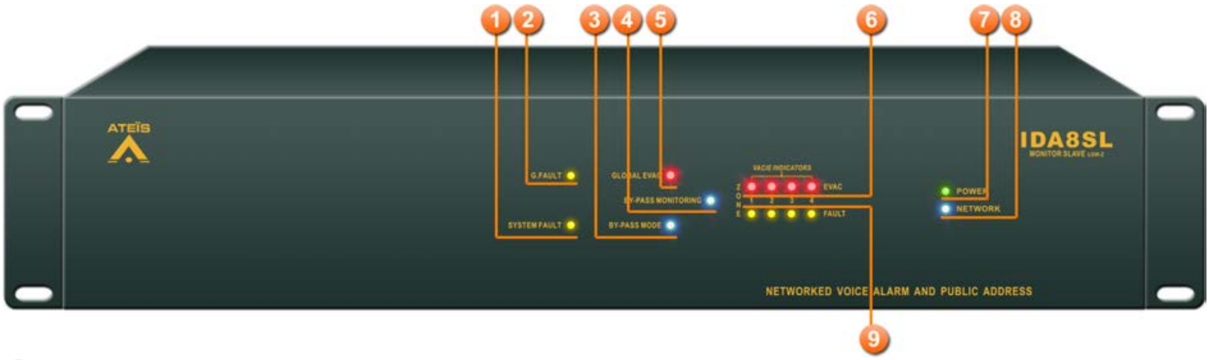


The IDA8SL Slave units provide audio output expansion of the IDA8 Systems using a secured 48-channel audio and data network over CAT5 or fiber optic. Each IDA8SL expands the IDA8 System with an additional 4 outputs, 2 additional security microphone consoles PSS AS and security programmable switching contacts.

The Network cards that comes with the unit provides a redundant 48 channels audio and data connection, Ateïs-Net, between one controller and a maximum of 32 IDA8SX Slave units in one rack system.

The IDA8SL is used on PA/VA applications that need large amplifier power. The maximum power may reach 1000W(RMS). The connectors for audio are Speakon style, which is mostly used in professional audio systems for connecting loudspeakers to amplifiers.

4.1.3.2 Front Panel



1 System Fault LED

This LED lights on when a system fault fault is detected.

2 G.Fault LED

This LED lights on when a global fault is detected.

3 Bypass Mode LED

To indicate IDA8C is in bypass mode or not.

4 Bypass Monitoring LED

Show the monitoring function is enabled or not. This LED lights on if monitoring of IDA8 is disabled by user.

5 Global EVAC LED

To indicate system is in EVAC paging state or not. This LED lights on if any IDA8 device over Ateis Net do an EVAC paging. See the topic of Network Paging Component to learn more about how to config a EVAC paging.

6 Zone EVAC LEDs

To show the audio channel is in EVAC paging or not. Each LED is correspond to a output channel of Network Paging component, for example, 1st LED is for pin M1, 2nd LED for pin M2 and so on.

7 Power LED

This LED lights on when IDA8SL is power on.

8 Network LED

Show the status of Ateis Net. This LED lights on if more than two deployed IDA8 devices in Ateis Net.

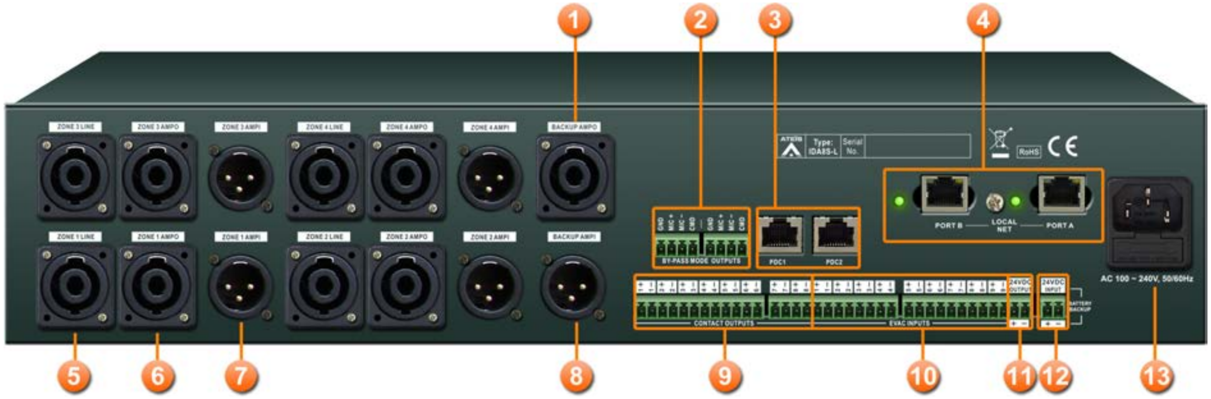
9 Zone Fault LEDs

To show fault status of zone. This LED light on if one of following faults is detected:

- Normal AMP Error
- Line A Error

- Line B Error
- Backup AMP Error
- AMP Line Leakage Error

4.1.3.3 Rear Panel



1 Backup Amplifier Output

This connector connects to the output of the backup amplifier for receiving the 100V audio signal powered by the amplifier.

2 Bypass Mode Outputs

This port to share the bypass mode microphone signals through the IDA8C-IDA8S network(only needed when using fiber optic network).

3 PDC (Peripherals Device Controller) Connectors

Two RJ-45 connectors to connect consoles or peripheral devices. For examples, PSS AS, URC AS, PPM AS, ... are connect to IDA8SAB via this connector.

4 Local Ateis Net Connectors

Optional card to build a local dedicated IDA8C-IDA8S network.

5 Zone1 Speaker Line Output

This connector connects to the loudspeaker of zone1.

6 Zone1 Amplifier Output

This connector connects to the output of the zone1 amplifier for receiving the 100V audio signal powered by the amplifier.

7 Zone1 Amplifier Input

This connector connects to the input of the zone1 amplifier for transmitting 0dB audio signal to the amplifier.

8 Backup Amplifier Input

This connector connects to the input of the backup amplifier for transmitting 0dB audio signal to the

amplifier.

9 Contact Outputs

8 logic outputs channels to close/open circuit for an external device, this contact is normally open.

10 Evacuation Inputs

9 evacuation contact inputs that allow the monitoring of external contact. They also can be used in UGA mode, trigger by a voltage polarization change.

11 24V DC Output

This connector supplies a 24VDC source.

12 24V DC Input

Main 24VDC power supply connector.

13 AC Power Socket

Main 110~240 V 1.2A, 47~63Hz AC power input with fuse. If 24V DC and AC power input at the same time, IDA8C will use AC power, and switch to DC power if there is no AC power input.

Fuse Rating: 1.6A

4.1.3.4 Characteristics

❖ Case

- Dimension = 436mm (W) x 300mm (L) x 88mm (H).
- Weight = 5Kg.
- Color = RAL7016.

❖ Power

Item	Voltage	Current Consumption	Comment
AC Input	100V~240V	1.2A@100V, 0.5A@240V	Frequency:47Hz~63Hz
DC Input	18V~26V, Typical 24V	2A	-
DC Output	18V~26V, Typical 24V	0.5A	-

AC Maximum Consumption = 45 W.

❖ Amplifier Zone Outputs

- Maximum level = 14 dB
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Speaker line/Amplifier

- Maximum Power = 100W(RMS).
- Maximum Power(Siren + Message) = 1500W.

❖Configurable Audio Output

- Maximum level = 14 dBu
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

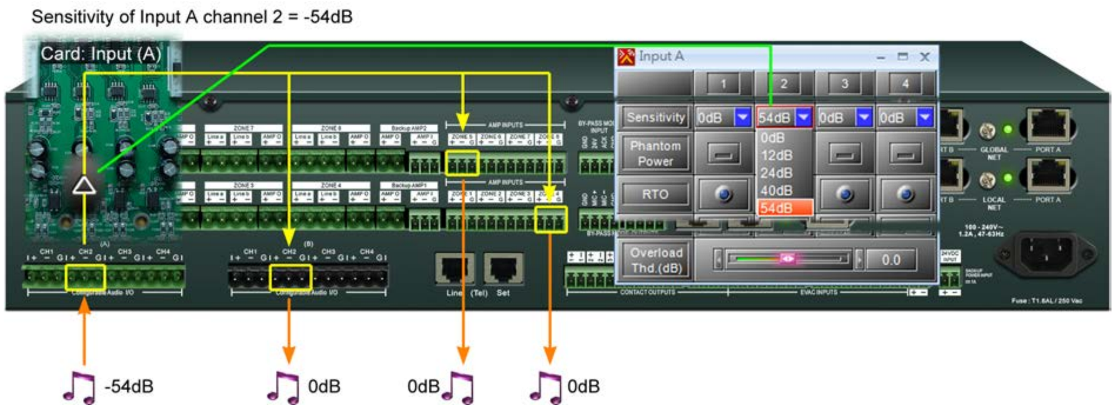
- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Configurable Audio Input

- Sensitivity

  For each channel of configurable audio input card, there are five levels to gain the audio signal. They are sensitivity values of 0dBu, -12dbBu, -24dBu, -40dBu and -54dBu respectively. The meaning of sensitivity value is how large the gain for an audio source, and to amplify the giving minus input source to 0dBu, the figure below is an example when the sensitivity value is set to -54dBu, and input source is -54dBu, after the gaining circuit of audio card, you'll get the 0dBu of output.



- Maximum level = 14 dBu
- Input Impedance = 10k Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.

❖PDC

- Maximum Output Level = 10 dBu.
- Output Impedance = 300 Ohm.
- THD @ 1kHz at output < 0.02.
- Bandwidth @ -3dB at output = 20Hz ~ 20kHz.
- Noise @ 22Hz ~ 22kHz = -85dBu.
- Maximum Input Level = 16 dBu.
- Input Impedance = 11k Ohm.
- THD @ 1kHz at input < 0.02
- Bandwidth @ -3dB at input < 20Hz ~ 22kHz.
- RS232 voltages = -6.5/+6.5

❖Evacuation Inputs(Contact Mode)

- Bias voltage:

Item	Minimum	Maximum	Unit
Voltage	-	5	VDC

- Monitoring resistor = 4.7k Ohm.

❖Evacuation Inputs(Voltage Mode)

On Voltage:

Item	Minimum	Maximum	Unit
Voltage	18	72	VDC

❖Contact Outputs + EVAC, Fault State Outputs

Item	Minimum	Maximum	Unit
Voltage	-	100	VDC
Current	-	0.5	ADC

❖Working Temperature.

- 0°C ~ 40°C

4.1.3.5 3rd Party Control

The following table list 3rd Party Control protocols IDA8C Supported.

Protocol	Interface		Function
	Connection	Settings	
Ateis 3rd Party Protocol	Ethernet(UDP)	UDP Port = 19761	• Read/Write value of parameters. • Stepped adjustment for parameters.
Modbus	PDC(RS485)	PDC Port 1~ 2	• Level control • Paging • Read Monitoring Status • Read Evacuation Status • Play Message • Logic I/O Control

4.1.3.6 Peripherals

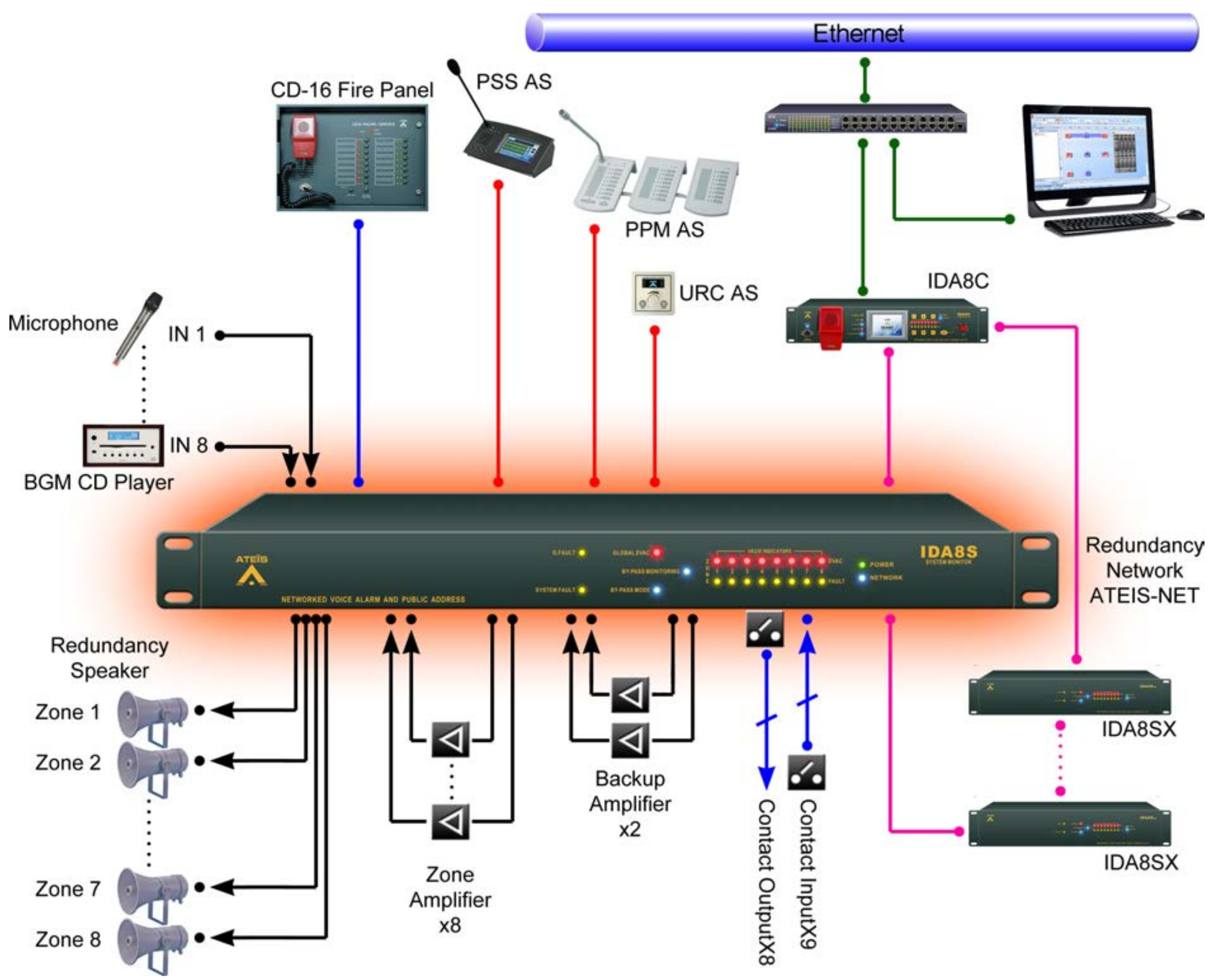
Following table is the peripherals supported by IDA8SL:

Device	Connection	Max. Num	Function
Fireman MIC	Fireman MIC Input	1	Paging where operator is close to IDA8C.
PPM AS	PDC	31/PDC	Remote console for with paging ability.
PSS AS	PDC	1/PDC	Remote console for Paging, Event Triggering(Element Control, Master/Sub Preset Control).
URC AS	PDC	1/PDC	Remote console for Master/Sub Preset Controlling, Element Controlling.
URGP	PDC	1/PDC	Evacuation Input extension.
DNM	PDC	1/PDC	Auto noise gain for audio signals.
PPM-IT5	Ethernet	1 Active Over Eth	Paging, Element Control, Master/Sub Preset Control.
URC200 TPC	Ethernet	Eth. Limit	Parameters Control, Master/Sub Preset Control.
CD8	PDC	31/PDC	Wall mounted cabinet remote paging console with 8 buttons/ zones.
CD16	PDC	31/PDC	Wall mounted cabinet remote paging console with 16 buttons/

Device	Connection	Max. Num	Function
			zones.
CD Touch	PDC	1/PDC	Wall mounted cabinet remote paging console with touch screen and Fireman Microphone.
PCP	PDC	31/PDC	Wall mounted cabinet remote paging console with telephone styled microphone.
CDPA	PDC	31/PDC	Wall mounted cabinet remote paging console with 24 buttons/ zones and 2 extra selectable microphone sources.
PSC	PDC	31/PDC	A grouped console is comprised of a pad with monitoring speaker, a pad with a gooseneck and 8 buttons/zones, two pads with 8 buttons/zones.
Deskpad	PDC	1/PDC	A remote dialer for making telephone calls via IDA8's telephone hardware.
Deskpad Box	PDC	1/PDC	It's a RF transceiver to communication between Deskpad and IDA8, use it, Deskpad can working without physcial connection.

4.1.4 IDA8S

4.1.4.1 Overview

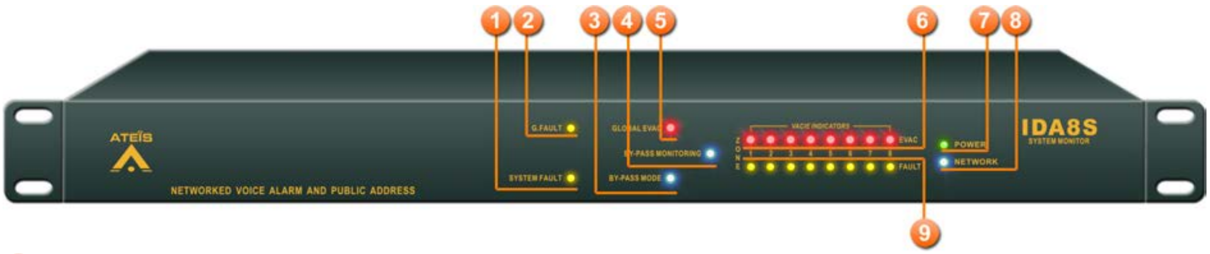


The IDA8S Slave units provide audio in and output expansion of the IDA8 Systems using a secured 48-

channel audio and data network over CAT5 or fiber optic. Each IDA8S expands the IDA8 System with an additional 8 outputs, 2 additional security microphone consoles PSS AS and security programmable switching contacts.

The Network cards that comes with the unit provides a redundant 48 channels audio and data connection, Ateïs-Net, between one controller and a maximum of 32 IDA8SX Slave units in one rack system.

4.1.4.2 Front Panel



1 System Fault LED

This LED lights on when a system fault fault is detected.

2 G.Fault LED

This LED lights on when a global fault is detected.

3 Bypass Mode LED

To indicate IDA8C is in bypass mode or not.

4 Bypass Monitoring LED

Show the monitoring function is enabled or not. This LED lights on if monitoring of IDA8 is disabled by user.

5 Global EVAC LED

To indicate system is in EVAC paging state or not. This LED lights on if any IDA8 device over Ateis Net do an EVAC paging. See the topic of Network Paging Component to learn more about how to config a EVAC paging.

6 Zone EVAC LEDs

To show the audio channel is in EVAC paging or not. Each LED is correspond to a output channel of Network Paging component, for example, 1st LED is for pin M1, 2nd LED for pin M2 and so on.

7 Power LED

This LED lights on when IDA8S is power on.

8 Network LED

Show the status of Ateis Net. This LED lights on if more than two deployed IDA8 devices in Ateis Net.

9 Zone Fault LEDs

To show fault status of zone. This LED light on if one of following faults is detected:

- Normal AMP Error
- Line Speaker Error
- Backup AMP Error
- AMP Line Leakage Error

4.1.4.3 Rear Panel

1 Speaker Zone Outputs

There are 8 zones for speaker & amplifier connection. each zone consists of following connectors (from left to right):

- connector of line speaker
- connector of 100V audio coming from amplifier.

2 Backup Amplifier I/Os

Two backup amplifier connectors included 0 dB to amplifiers and 100V return from amplifiers.

3 Amplifier Zone Outputs

8 Zone 0dB output to amplifiers.

4 Bypass Mode Outputs

This port to share the bypass mode microphone signals through the IDA8C-IDA8S network(only needed when using fiber optic network).

5 PDC(Peripherals Device Controller) Connectors

Two RJ-45 connectors to connect consoles or peripheral devices. For examples, PSS AS, URC AS, PPM AS, ... are connect to IDA8S via this connector.

6 Local Ateis Net Connectors

Optional card to build a local dedicated IDA8C-IDA8S network.

7 Configurable Audio I/Os

Two configurable 0dB audio I/O port A and B. Each port is available to assemble an audio card. There are 4 channels on each audio I/O card.

8 Contact Outputs

8 logic outputs channels to close/open circuit for an external device, this contact is normally open.

9 Evacuation Inputs

9 evacuation contact inputs that allow the monitoring of external contact. They also can be used in UGA mode, trigger by a voltage polarization change.

10 24V DC Output

This connector supplies a 24VDC source.

11 24V DC Input

Main 24VDC power supply connector.

4.1.4.4 Characteristics

❖ Case

- Dimension = 436mm (W) x 285mm (L) x 44mm (H).
- Weight = 4.2Kg.
- Color = RAL7016.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	2A	-
DC Output	18V~26V, Typical 24V	0.5A	-

- AC Maximum Consumption = 45 W.

❖ Amplifier Zone Outputs

- Maximum level = 14 dB
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Speaker line/Amplifier

Maximum Power = 700W(RMS).

Maximum Power(Siren + Message) = 1000W.

❖Configurable Audio Output

- Maximum level = 14 dBu
- Output Impedance = 50 Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.
- EIN(Equivalent Input Noise)

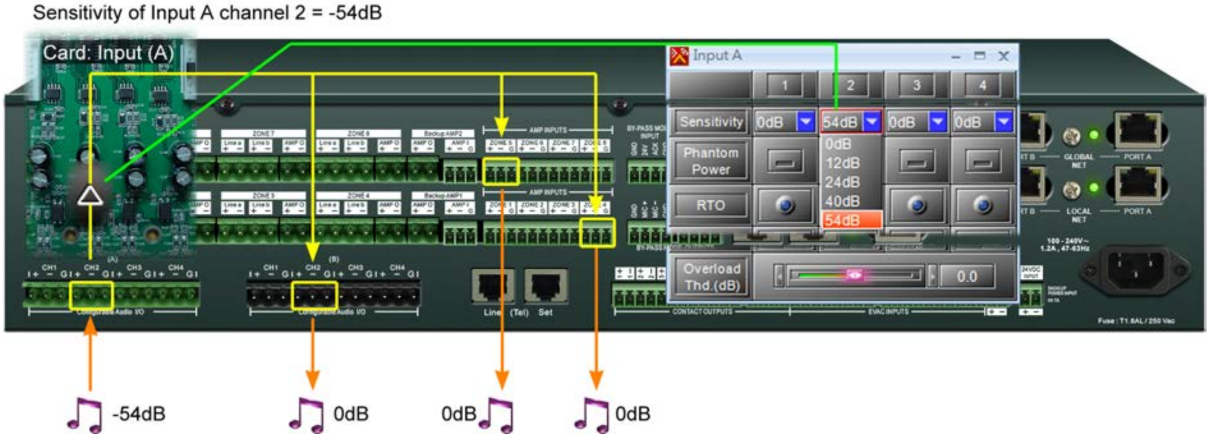
Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	-81	dBu	20~20kHz@150Ù
-12dBu	-	-93	dBu	20~20kHz@150Ù
-24dBu	-	-104	dBu	20~20kHz@150Ù
-40dBu	-	-112	dBu	20~20kHz@150Ù
-54dBu	-	-115	dBu	20~20kHz@150Ù

❖Configurable Audio Input

- Sensitivity

For each channel of configurable audio input card, there are five levels to gain the audio signal. They are sensitivity values of 0dBu, -12dbBu, -24dBu, -40dBu and -54dBu respectively. The meaning of sensitivity value is how large the gain for an audio source, and to amplify the giving minus input source to 0dBu, the figure below is an example when the sensitivity value is set to -54dBu, and input source is -54dBu, after the gaining circuit of audio card, you'll get the 0dBu of output.

Sensitivity of Input A channel 2 = -54dB



- Maximum level = 14 dBu
- Input Impedance = 10k Ohm
- THD+N(Total Harmonic Distortion plus Noise)

Sensitivity	Minimum	Maximum	Unit	Frequency
0dBu	-	0.04	%	20~20kHz@+4dBu
-12dBu	-	0.06	%	20~20kHz@-2dBu
-24dBu	-	0.06	%	20~20kHz@-14dBu
-40dBu	-	0.06	%	20~20kHz@-30dBu
-54dBu	-	0.06	%	20~20kHz@-44dBu

- Bandwidth = 20Hz ~ 20kHz.

❖PDC

- Maximum Output Level = 10 dBu.
- Output Impedance = 300 Ohm.
- THD @ 1kHz at output < 0.02.
- Bandwidth @ -3dB at output = 20Hz ~ 20kHz.
- Noise @ 22Hz ~ 22kHz = -85dBu.
- Maximum Input Level = 16 dBu.
- Input Impedance = 11k Ohm.
- THD @ 1kHz at input < 0.02
- Bandwidth @ -3dB at input < 20Hz ~ 22kHz.
- RS232 voltages = -6.5/+6.5

❖Evacuation Inputs(Contact Mode)

- Bias voltage:

Item	Minimum	Maximum	Unit
Voltage	-	5	VDC

- Monitoring resistor = 4.7k Ohm.

❖Evacuation Inputs(Voltage Mode)

On Voltage:

Item	Minimum	Maximum	Unit
Voltage	18	72	VDC

❖Contact Outputs + EVAC, Fault State Outputs

Item	Minimum	Maximum	Unit
Voltage	-	100	VDC
Current	-	0.5	ADC

❖Working Temperature.

- 0°C ~ 40°C

4.1.4.5 3rd Party Control

The following table list 3rd Party Control protocols IDA8C Supported.

Protocol	Interface		Function
	Connection	Settings	
Ateis 3rd Party Protocol	Ethernet(UDP)	UDP Port = 19761	• Read/Write value of parameters. • Stepped adjustment for parameters.
Modbus	PDC(RS485)	PDC Port 1~ 2	• Level control • Paging • Read Monitoring Status • Read Evacuation Status • Play Message • Logic I/O Control

4.1.4.6 Peripherals

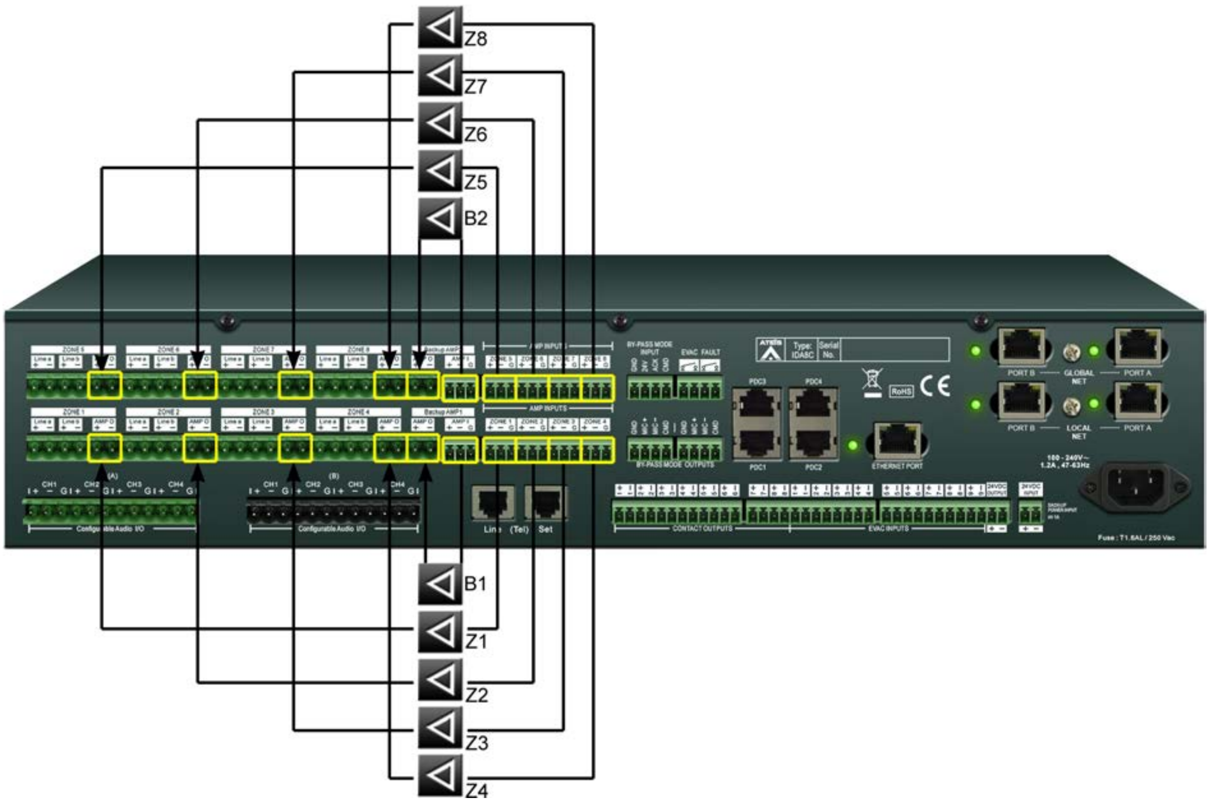
Following table is the peripherals supported by IDA8S:

Device	Connection	Max. Num	Function
Fireman MIC	Fireman MIC Input	1	Paging where operator is close to IDA8C.
PPM AS	PDC	31/PDC	Remote console for with paging ability.
PSS AS	PDC	1/PDC	Remote console for Paging, Event Triggering(Element Control, Master/Sub Preset Control).
URC AS	PDC	1/PDC	Remote console for Master/Sub Preset Controlling, Element Controlling.
URGP	PDC	1/PDC	Evacuation Input extension.
DNM	PDC	1/PDC	Auto noise gain for audio signals.
PPM-IT5	Ethernet	1 Active Over Eth	Paging, Element Control, Master/Sub Preset Control.
URC200 TPC	Ethernet	Eth. Limit	Parameters Control, Master/Sub Preset Control.
CD8	PDC	31/PDC	Wall mounted cabinet remote paging console with 8 buttons/ zones.
CD16	PDC	31/PDC	Wall mounted cabinet remote paging console with 16 buttons/ zones.
CD Touch	PDC	1/PDC	Wall mounted cabinet remote paging console with touch screen and Fireman Microphone.
PCP	PDC	31/PDC	Wall mounted cabinet remote paging console with telephone styled microphone.
CDPA	PDC	31/PDC	Wall mounted cabinet remote paging console with 24 buttons/ zones and 2 extra selectable microphone sources.
PSC	PDC	31/PDC	A grouped console is comprised of a pad with monitoring speaker, a pad with a gooseneck and 8 buttons/zones, two pads with 8 buttons/zones.
Deskpad	PDC	1/PDC	A remote dialer for making telephone calls via IDA8's telephone hardware.
Deskpad	PDC	1/PDC	It's a RF transceiver to communication between Deskpad and

Device	Connection	Max. Num	Function
Box			IDA8, use it, Deskpad can working without physcial connection.

4.1.5 Amplifier Configuration

4.1.5.1 Basic Amplifier Connection

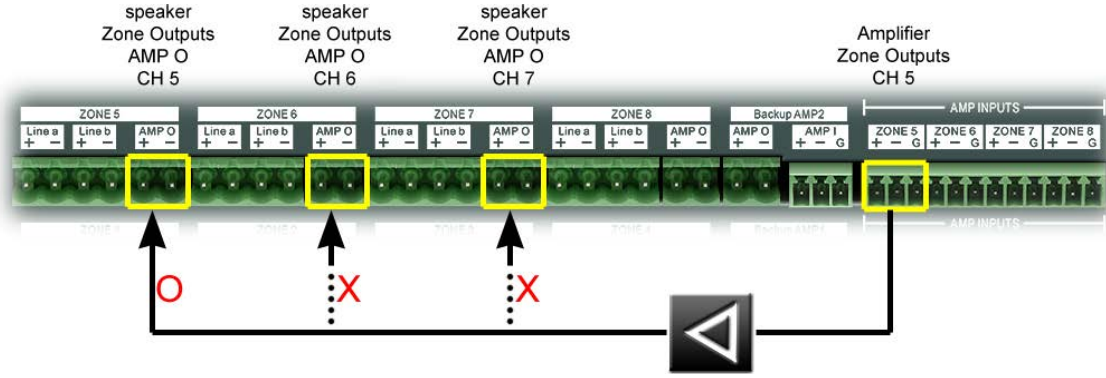


Basic Amplifier Connection

IDA8 System apply a flexible amplifier setup , the figure above is the basic setup of amplifier with IDA8. The figure show the connection between amplifiers and IDA8C. There two kinds of connections for amplifier:

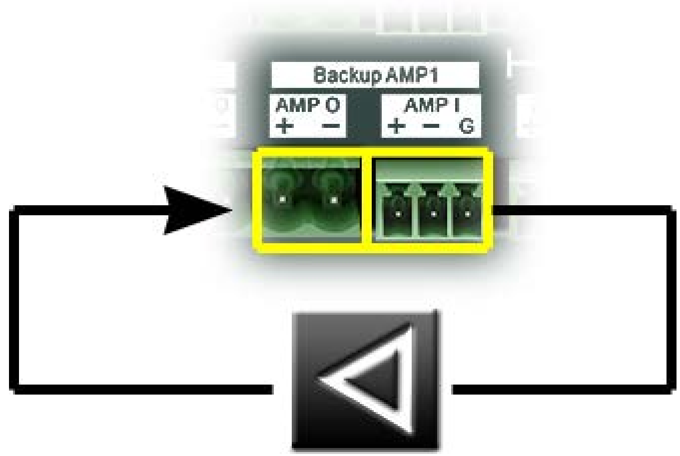
- Normal Amplifiers Connection

  The connection of amplifiers for speaker zone outputs. In above figure(Basic Amplifier Connection), Z1~Z8 are normal amplifiers. Follow the wires, audio signal start from amplifier zone outputs goes into amplifier's input, then gained by amplifier and input to AMPO connector of speaker line I/Os. For each channel of amplifier zone outputs should connect to it's correspond channel of speaker zone output's AMPO connector. For example, amplifier zone output 5 should connect to zone5 AMPO connector of speaker zone outputs, see the below figure.



- Backup Amplifiers Connection

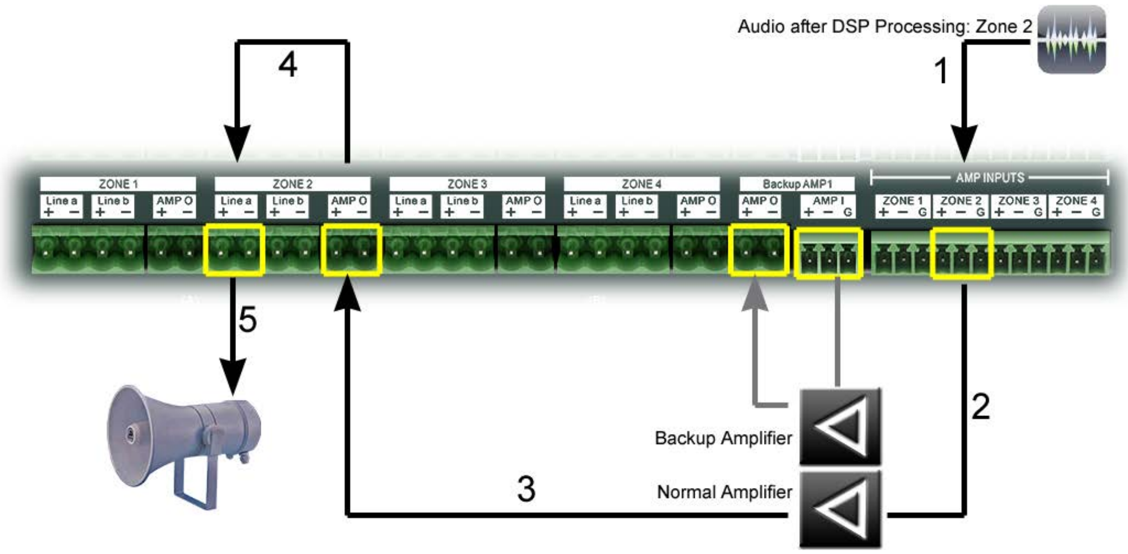
The connection of amplifiers for backup. In above figure(Basic Amplifier Connection), B1, B2 are backup amplifiers. There two connectors for each backup amplifier. Follow the wire, audio signal start from the AMP I connector, goes into the amplifier's input, then gained by amplifier and input to AMP O connector. The following figure is an example of backup amplifier connection.



4.1.5.2 Amplifier Backup

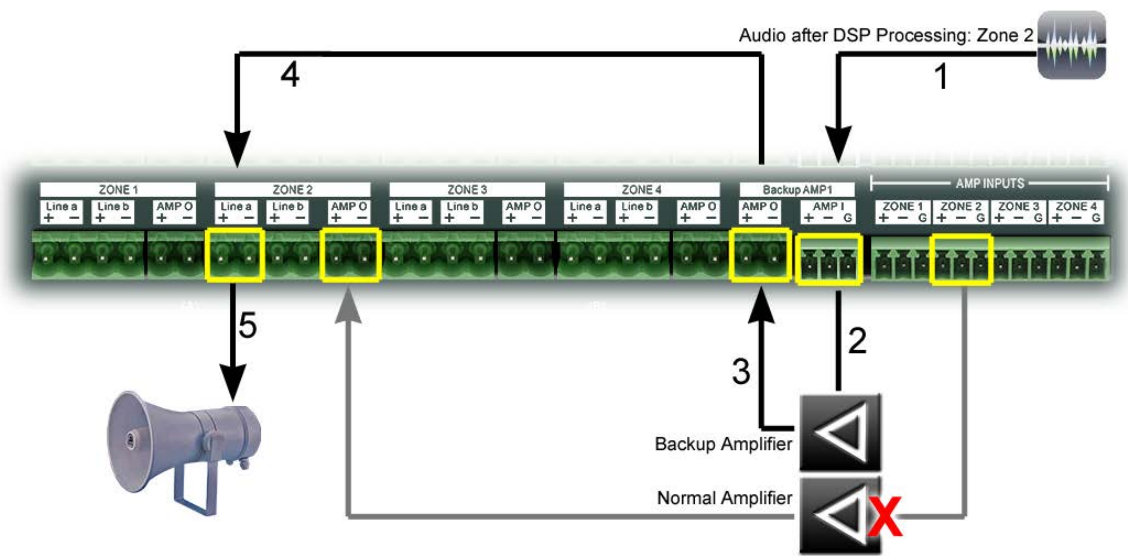
❖Basic Concept

IDA8 system apply a flexible mechanism for amplifier backup. Before dig deeper inside the rule of amplifier backup, let's understand the basic concept of it. For most of case audio go through the normal amplifier, once the normal is failure IDA8 will switch to backup amplifier to make sure the system still working. At this time, the technician should analysis the problem, of course, to replace the bad amplifier with good one. The figure(Audio Routing without Backup Amplifier) show the general case of audio routing using normal amplifier, the number in figure is the sequence of audio path.



Audio Routing without Backup Amplifier

The figure(Audio Routing with Backup Amplifier) shows the amplifier backup case for audio routing. When a error of normal amplifier is detected by IDA8X, IDA8X will change audio path from normal amplifier to backup amplifier, and generate a fault to warning user that there was an amplifier fault detected by system, please take care about it.

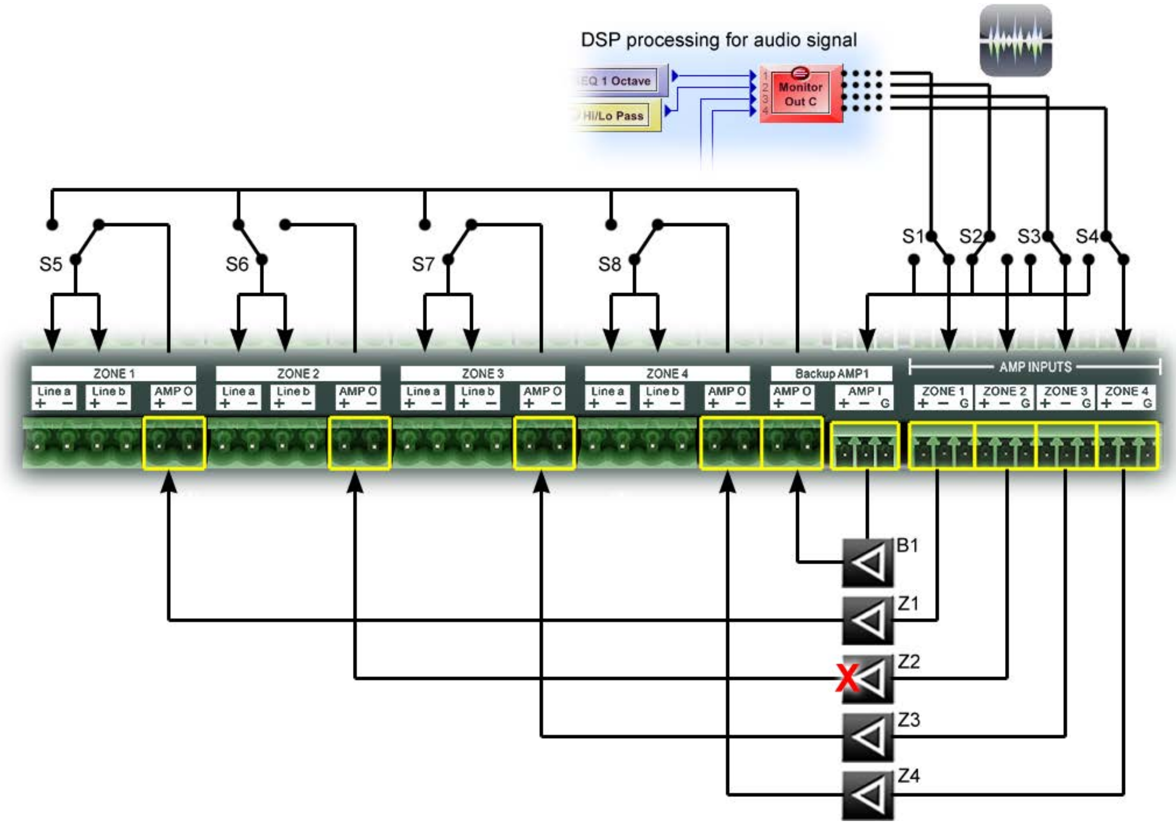


Audio Routing with Backup Amplifier

❖ Share the Backup Amplifier

For the most of case, two amplifier should not be failure at the same time, it means the possibility of more than two amplifier failure is pretty low. It is no necessary to have a backup amplifier for each zone, because most of time the back up amplifiers are just wait for the case of normal amplifier fail. If one of normal amplifier fail, one backup amplifier can take over immediately, at the mean time, technical stuff should switch the bad amplifier at a short time. During the time of amplifier backup, It is almost not possible to have another amplifier fail. Base on this concept, IDA8 system provide a flexible mechanism of amplifier backup. One backup amplifier can service multiple zones. In preview figures it demonstrate how a backup amplifier work when normal amplifier of zone 2 is failure, but the

backup amplifier can serve for zone 1 to zone 4 actually. The below figure shows the concept of that.

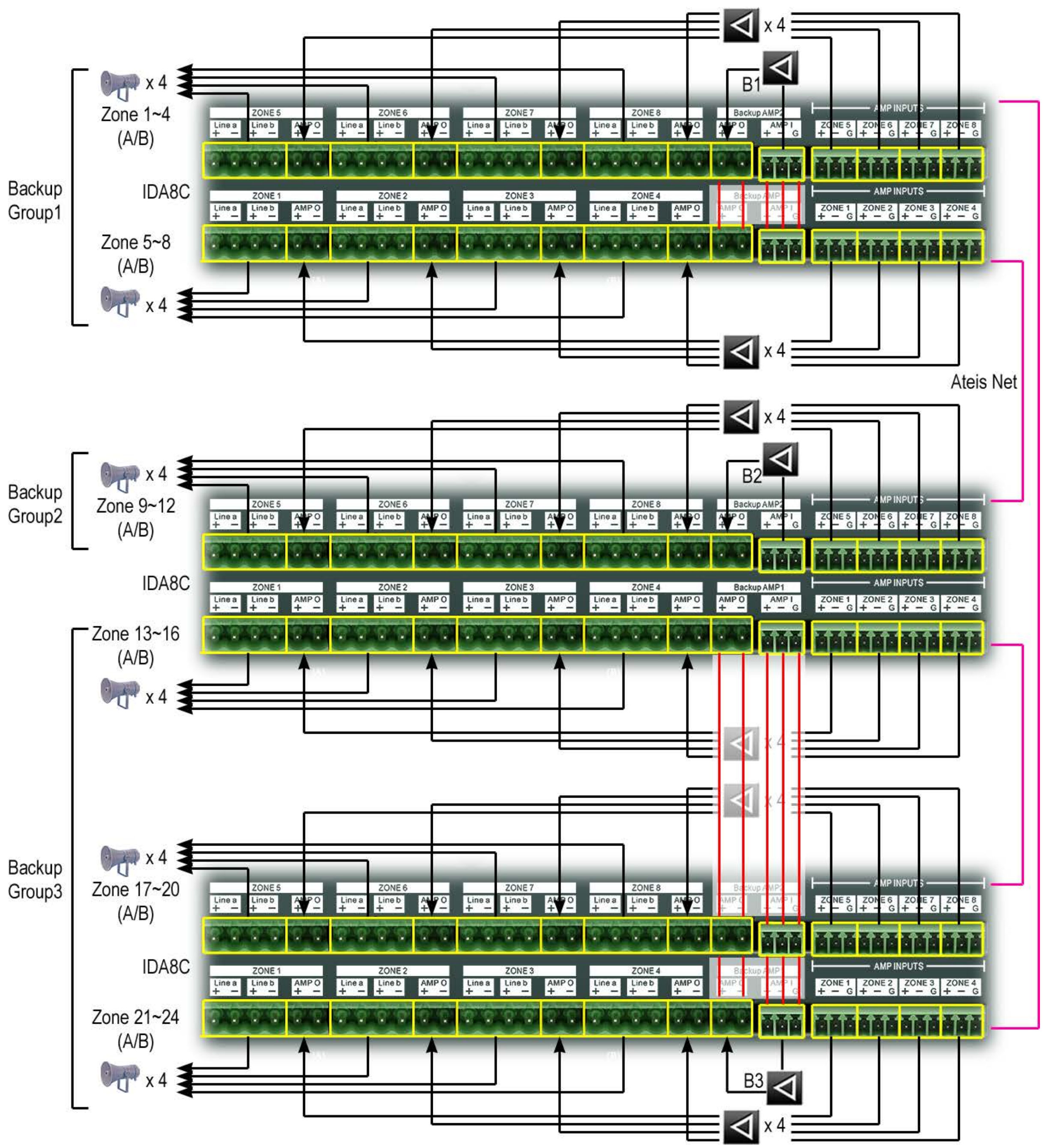


Share the Backup Amplifier

In the figure, Z1~Z4 represent the normal amplifier for Z1~Z4, B1 is backup amplifier. S1~S4 are switches to select path of audio signal coming from DSP processing. S5~S8 are switches to select path going to Line A/B output. B1 can service for zone1 to zone 4 by controlling switches S1~S8. In example, amplifier Z2 is fail, IDA8 control S6 and S2 to change audio path from normal amplifier Z2 to backup amplifier B1. The behavior of other channels are the same as channel2. A backup amplifier can't serve more than one zone at the same time. In such limitation, for example, if amplifier Z3 is fail, then there is no available amplifier for backup.

❖Advance Backup Amplifier Setup

There are more cost down solution of IDA8 backup amplifier setup. It is possible to let one backup amplifier serve more than four zones. See the below figure, an Ateis Net system is consist of one IDA8C and two IDA8Ss, there are 24 zones in the system divide into three backup amplifier groups. For each group, there is an backup amplifier to redundant if any of normal amplifier in the group is failure. To group zones, you need to wiring AMPI and AMPO between backup amplifier connectors. red wires show the wire connections to group zones.



Advance Backup Amplifier Setup

There are three backup amplifier groups of above figure:

- Backup Group1

  Amplifier B1 serves Zone1 ~ Zone8.

- Backup Group2

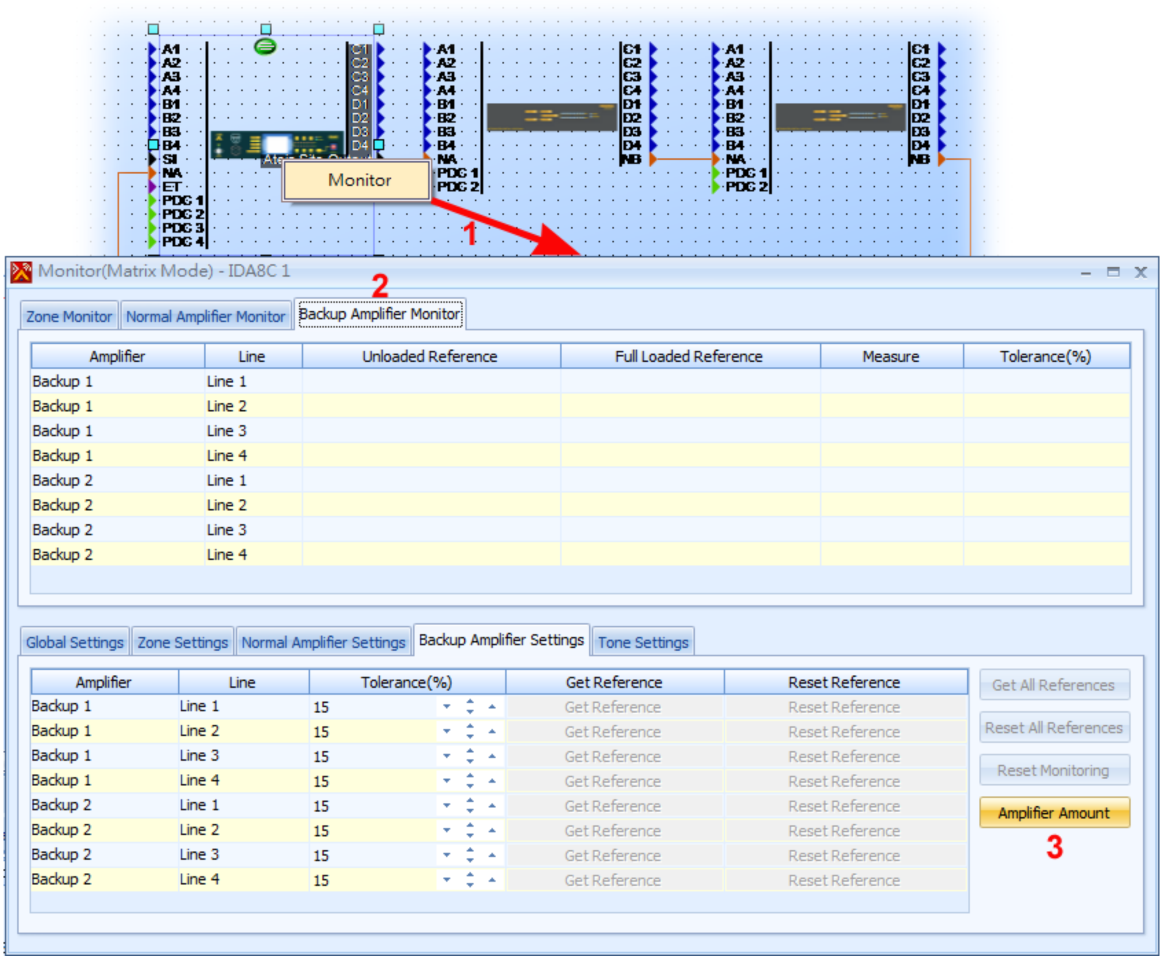
  Amplifier B2 serves Zone9 ~ Zone12.

- Backup Group3

Amplifier B3 serves Zone 13 ~ zone24.

❖ Software settings for backup amplifier

It is necessary to have software settings to make backup amplifier working.



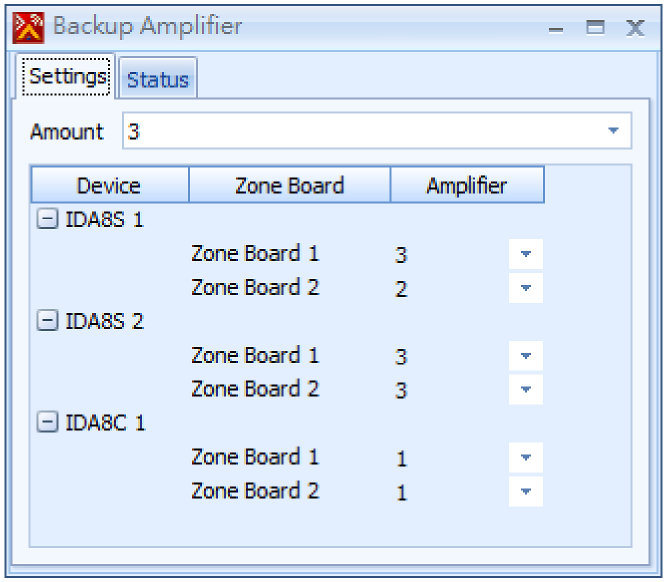
Numbers in figure show steps to open backup amplifier settings window:

1. Right click mouse on IDA8C1 block in devices window, then click item [Monitor] to open window "Monitor(Matrix Mode) - IDA8C1".

2. Select tab [Backup Amplifier Monitor].

3. Click button [Amplifier Amount]

Below figure show the software setup window for the backup configuration in Advance Backup Amplifier Setup :



The meaning of parameters list below:

- Amount

  Specify the number of backup amplifiers in system. the list content of combo box in amplifier depends on this value. For example, if Amount = 3, then the list content should be none, 1, 2, 3.

- Field [Device] of Grid

  List devices within the same Ateis Net.

- Field [Zone Board] of Grid

  List zone boards of the device. zone board1 means zone1 to zone4, zone board2 means zone5 to zone8.

- Field [Amplifier] of Grid

  Specify the group no of zones. In the above figure, IDA8S1 zone board1(zone1~zone4 of IDA8S1), IDA8S2 zone board1(zone1~4 of IDA8S2), IDA8S2 zone board2(zone 5~8 of IDA8S2) are the same value, i.e. those zones are the same group and use the same amplifier for backup.

4.1.6 Monitoring/Fault

For the security purpose, IDA8 is designed to monitor the equipment, when a error detected, a fault will be generated to inform user, then user can change the failure equipment quickly to make system working fine again.

There are two type of faults:

❖ System Fault

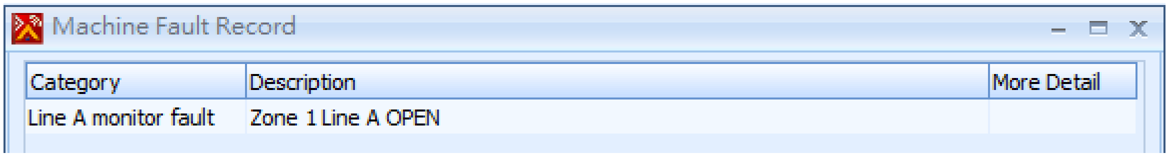
This kind of fault is about the internal error of IDA8C/S and most of case is hardware failure, it need to change board inside device to repair.

❖ Global Fault

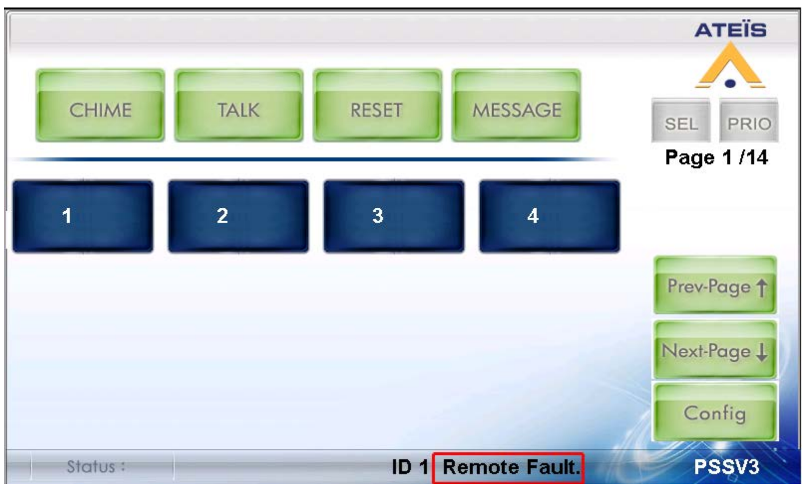
A global fault is mostly speak about the fault of equipment outside IDA8C/S.

When a fault is generated, the following things will be done by IDA8/S:

- System Fault LED is light on if the fault is a system fault, Fault. G.Fault LED is light on if the fault is a global fault.
- Fault State Output contact open.
- Log the fault
- A warning message for the fault will show in Touch Screen.
- When Ateis Studio is connect to IDA8C/S, in [Device Status] field of [Device Management] panel will show fault. Below is an example:



- There is a warning tone will be generate by monitoring speaker.
- A text message is showed in PSS touch screen, below figure is an example.



4.1.6.1 Table of System Faults

Name	Description
Flash Error	Nand flash inside IDA8C/S is error.
I2C Error	This is an internal error of I2C communication between micro controllers
Net Card Error	This is error about bad communication between main cpu in IDA8C/S and net card.
DSP Error	It's an internal error of communication between micro controllers inside devices.
Preset Table Error	The structure of configuration is not compatible with current firmware.
No Preset Table	There is no configuration inside device.
SPI Flash Error	An internal error of micro controller SPI communication.
Para. Table Error	Data error of table which store some important information of configuration.

Name	Description
Trans Error	An internal hw error of device.
Power Error	The power circuit inside device is error.
TEL Error	The communication between main controller unit and telephone is fail.
Main FPGA Error	Internal hw error.

4.1.6.1.1 Flash Error

❖ Fault Meaning

Nand flash inside IDA8C/S is error.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

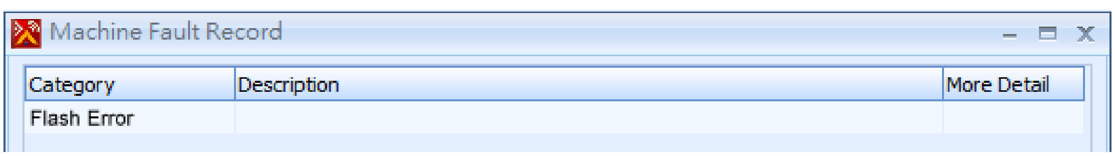
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Flash Error"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.2 I2C Error

❖ Fault Meaning

This is an internal error of I2C communication between micro controllers.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

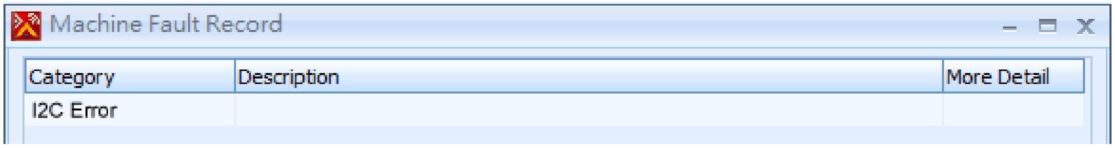
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

Text messages display at touch screen on front panel of IDA8C:

"I2C Error"

Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.3 Net Card Error

❖ Fault Meaning

This is error about bad communication between main cpu in IDA8C/S and net card.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

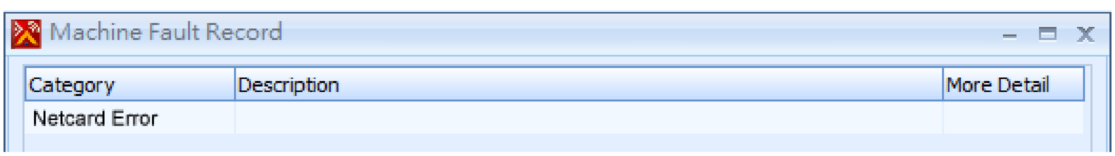
  - System Fault LED on front panel of IDA8C/S.
  - G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "NetCard Error"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.4 DSP Error

❖ Fault Meaning

It's an internal error of communication between micro controllers inside devices.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

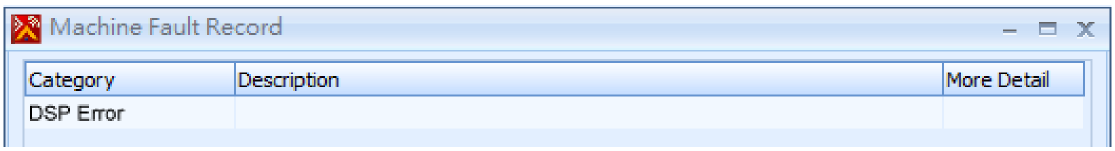
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "DSP Error"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.5 Preset Table Error

❖ Fault Meaning

The structure of configuration is not compatible with current firmware.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

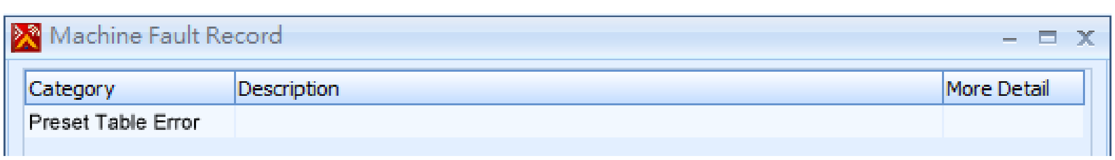
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Preset Table Err"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

1. Update firmware using newest version of Ateis Studio.

2. Store configuration. It the problem still not solved, please contact service.

4.1.6.1.6 No Preset Error

❖Fault Meaning

There is no configuration inside device.

❖Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

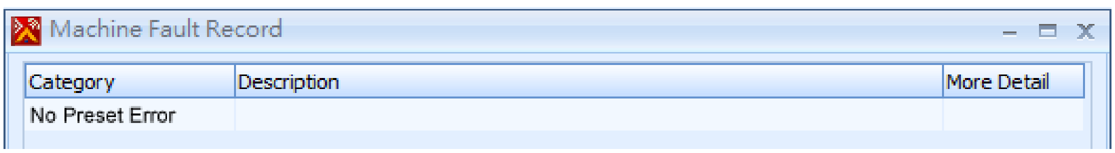
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "No Preset Table"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖Fault Handling

Store a configuration to device. If the error still not solved please contact service.

4.1.6.1.7 SPI Flash Error

❖Fault Meaning

An internal error of micro controller SPI communication.

❖Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

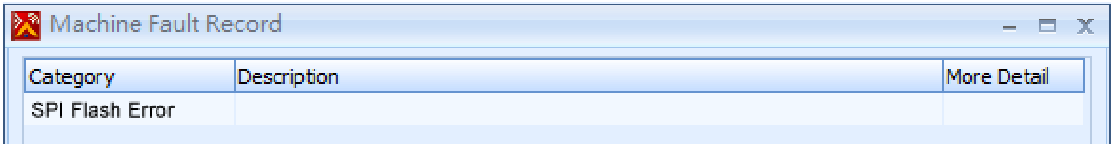
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "SPI Flash Error"

Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.8 Para. Table Error

❖ Fault Meaning

Data error of table which store some important information of configuration.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

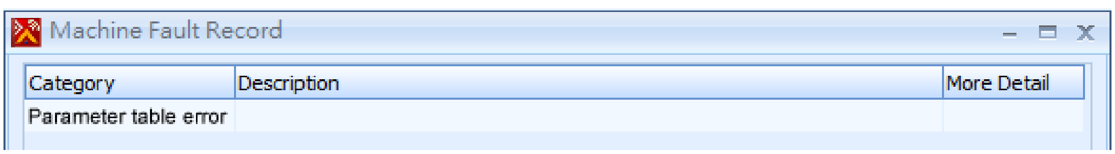
  - System Fault LED on front panel of IDA8C/S.
  - G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Para. Table Err"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Store a configuration to device. If the error still not solved please contact service.

4.1.6.1.9 Trans Error

❖ Fault Meaning

An internal hardware error of device.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

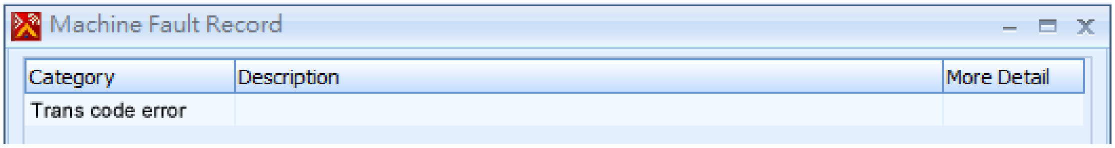
- ○ System Fault LED on front panel of IDA8C/S.
- ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Trans Error"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.10 Power Error

❖ Fault Meaning

The power circuit inside device is error.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

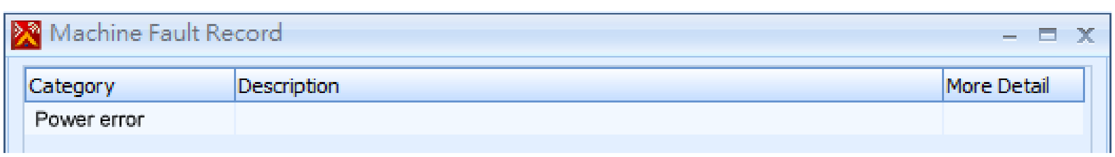
  - ○ System Fault LED on front panel of IDA8C/S.
  - ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Power Error"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.11 TEL Error

❖ Fault Meaning

The communication between main controller unit and telephone is fail.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

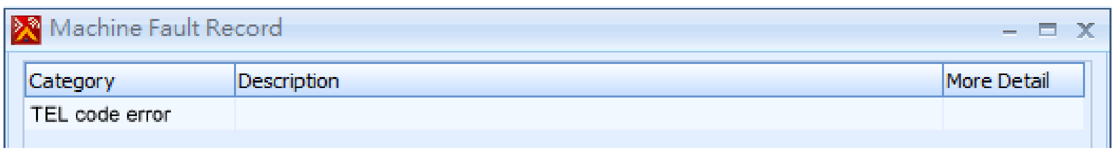
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "TEL Error"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.1.12 Main FPGA Error

❖ Fault Meaning

Internal hardware error.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ System Fault LED on front panel of IDA8C/S.

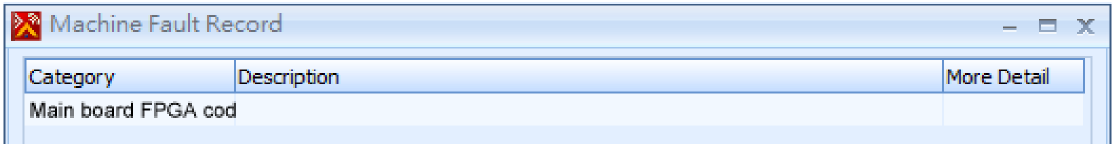
  ○ G. Fault LED on front panel of IDA8C/S.

- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Main FPGA Er"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Contact service to repair device.

4.1.6.2 Table of Global Faults

Name	Description
Normal/Backup AMP Error	1. Amplifier gain too high. 2. Amplifier gain too low. 3. Amplifier fail.
Line A/B Error	1. Line A/B Open, speaker is not connected to IDA8C/S. 2. Line A/B Bad Impedance, the impedance of speaker is changed over the tolerance. 3. Line A/B short, circuit outside the speaker zone output connector is shorted, no sound of if this fault happen, may damage the amplifier
AMP Line Leakage Error	The current between speakers and amplifiers is leakage.
EVAC Input Error	When evacuation input is set to contact mode, if the voltage of EVAC input is not in the range.
AteisNet Broken	This fault happens if one or more than one IDA8C or IDA8S over Ateis Net are lost connection.
Fireman Error	Fireman microphone is error.
VOX@NET Error	IDA8C/S can't communicate with VOX@NET server.
User Define Error	This fault happens if "Fault Definer" DSP component get a logic signal from input.
Remote Offline	The peripheral devices (PSS AS/PPM AS/URC AS/...) can't communication with IDA8C/S.
URGP Fault	URGP is fault
Remote Fault	The hw error(microphone/speaker on PSS) of peripheral devices,

4.1.6.2.1 Normal/Backup Amplifier Error

❖ Fault Meaning

- Amplifier gain is not high.
- Amplifier gain is too low.
- Amplifier is fail.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ G. Fault LED on front panel of IDA8C/S.

- Zone Fault LEDs on front panel of IDA8C/S.

- Log
  - Amplifier Gain Too High

    Text messages display at touch screen on front panel of IDA8C:

    "Normal AMP Error", where Normal AMP can be Backup AMP.

    "Z1 Normal AMP Too High", where Z1 can be any zone, Normal AMP can be Backup AMP.

    Text message display at window [Machine Fault Record] of Ateis Studio:


Machine Fault Record

Category	Description	More Detail
Normal AMP monitor fault	Zone 1 Too High	

  - Amplifier Gain Too Low

    Text messages display at touch screen on front panel of IDA8C:

    "Normal AMP Error", where Normal AMP can be Backup AMP.

    "Z1 Normal AMP Too Low", where Z1 can be any zone, Normal AMP can be Backup AMP.

    Text message display at window [Machine Fault Record] of Ateis Studio:


Machine Fault Record

Category	Description	More Detail
Normal AMP monitor fault	Zone 1 Too Low	

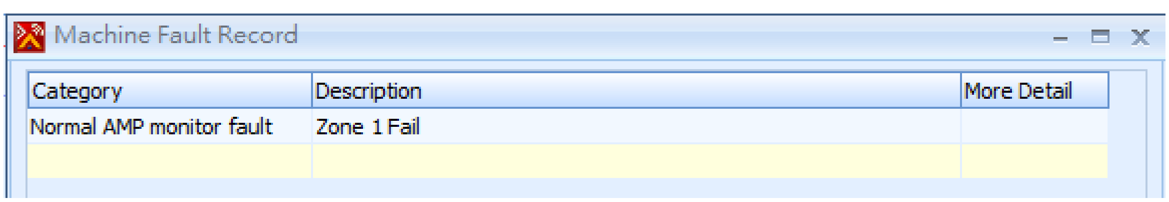
  - Amplifier Fail

    Text messages display at touch screen on front panel of IDA8C:

    "Normal AMP Error", where Normal AMP can be Backup AMP.

    "Z1 Normal AMP Fail", where Z1 can be any zone, Normal AMP can be Backup AMP.

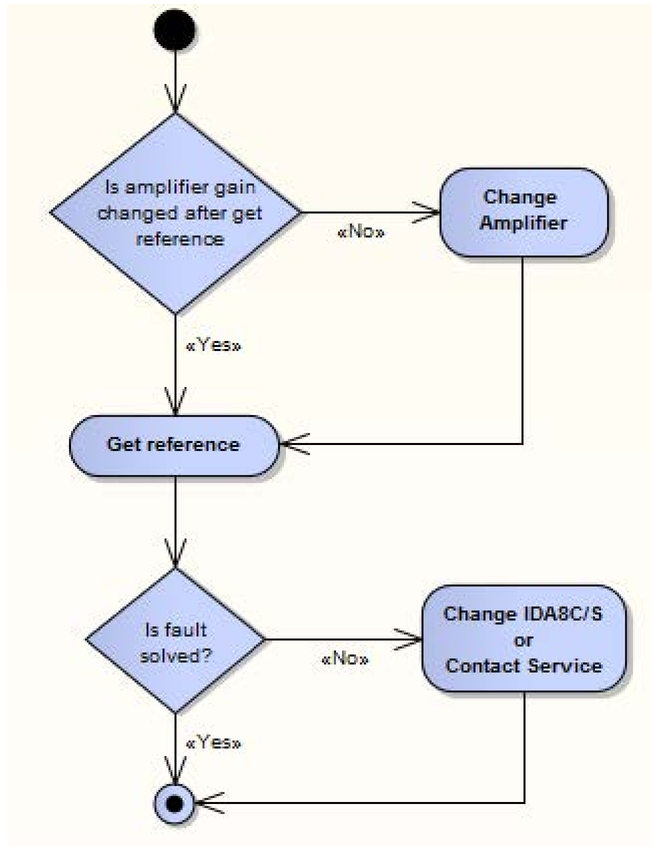
    Text message display at window [Machine Fault Record] of Ateis Studio:


Machine Fault Record

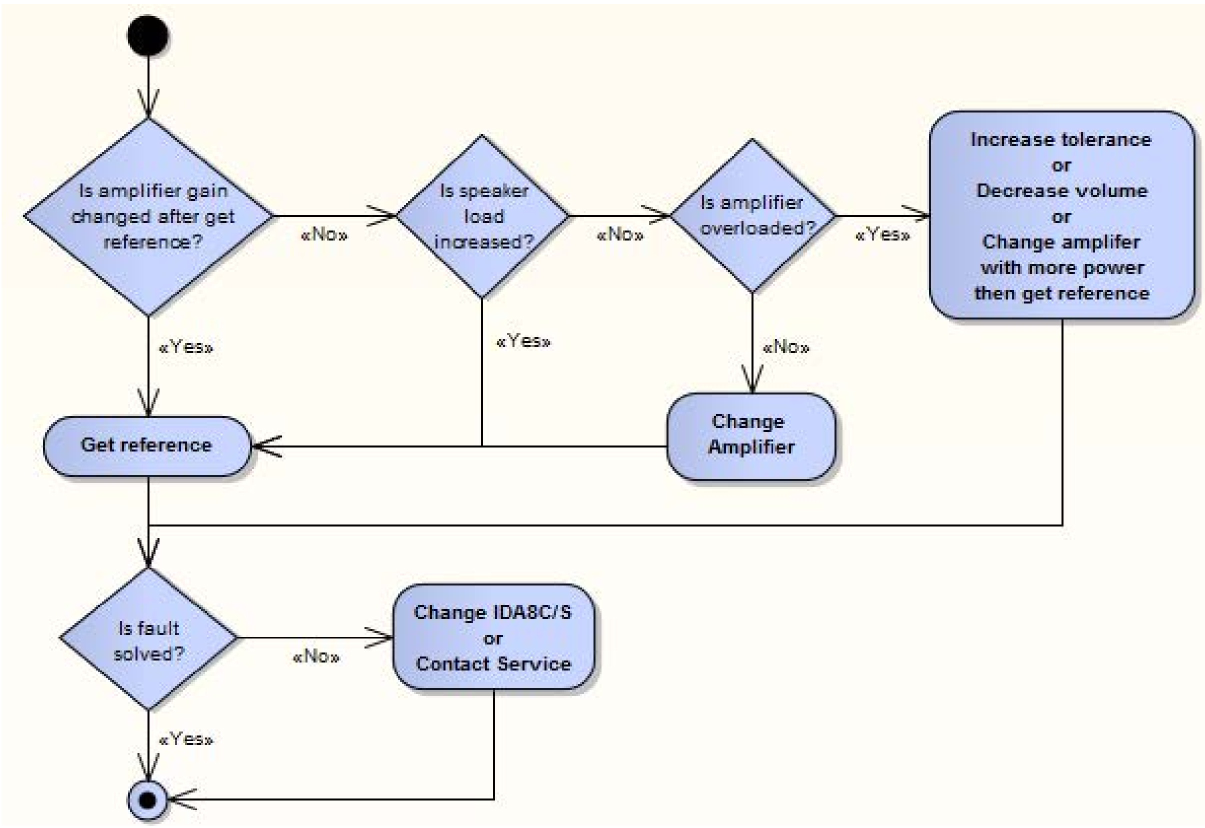
Category	Description	More Detail
Normal AMP monitor fault	Zone 1 Fail	

- Buzzer sound output to monitoring speaker on front panel of IDA8C.

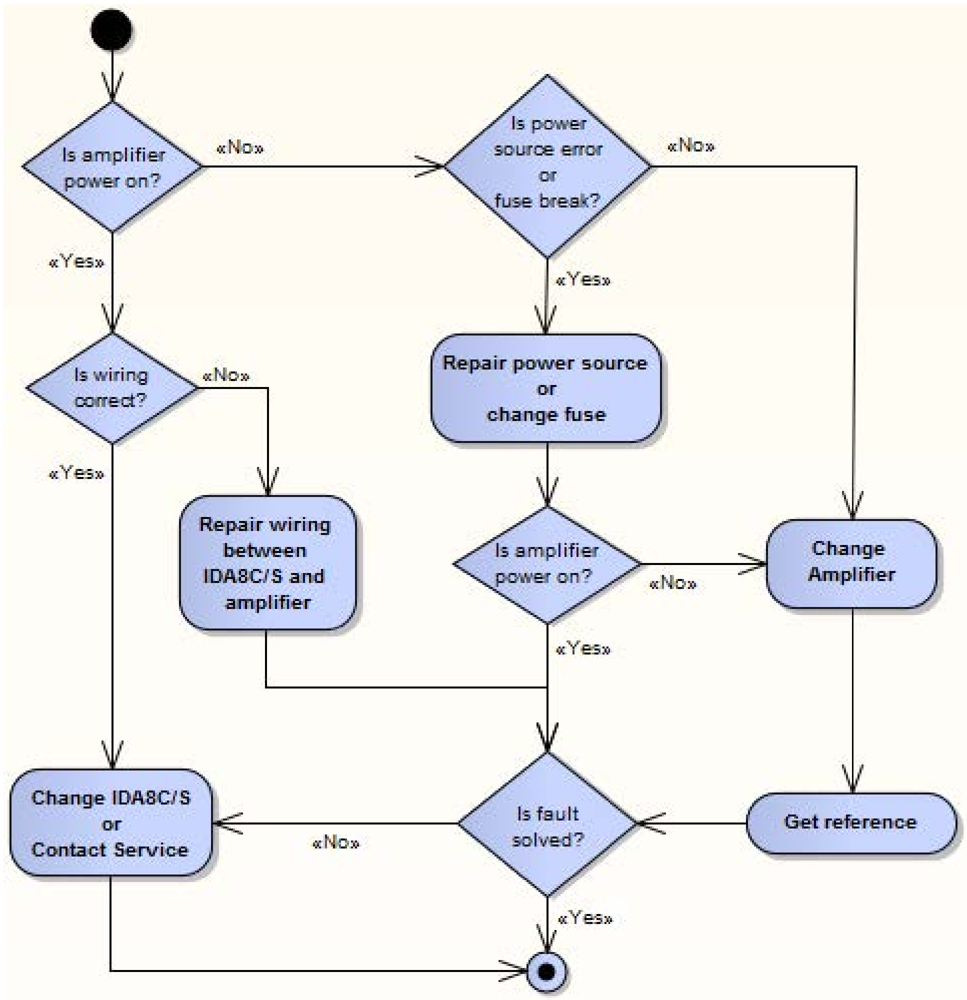
❖ Fault Handling



Amplifier Gain Too High



Amplifier Gain Too Low



Amplifier Fail

4.1.6.2.2 Line A/B Error

❖ Fault Meaning

This error is mainly for loudspeaker connected to IDA8C/S's speaker zone output connectors, there are three kind of faults:

- Line A/B Open

  If the value of Measure (ohm) is greater than 4000 ohm, this fault can be recognized by IDA8C/S, In the field of [Measure(ohm)] shows "OPEN".

- Line A/B Bad Impedance

  If the value of Measure (ohm) is greater than Reference A&B (ohm) + Tolerance (%), or value of Measure (ohm) is less than Reference A&B (ohm) - Tolerance (%), this fault can be recognized by IDA8C/S. In the field of [Diff(%)] shows the percentage of difference, in above figure, the difference percentage of zone 4 is 78.24 which is great than the tolerance 15%, a Line A/B bad impedance is recognized.

- Line A/B Short

The voltage of pin +, - is monitored by IDA8C/S, if the voltage is zero, it means there is a short of circuit outside IDA8C/S, if this fault is detected, the internal relay of zone will be open to stop audio signal output to speaker. User need to do a Reset Monitoring described in preview part of this topic to switch the status back, i.e. make audio output to speaker and enable monitoring of short again. A string "S.C." is showed in field [Measure(ohm)]

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ G. Fault LED on front panel of IDA8C/S.

  ○ Zone Fault LEDs on front panel of IDA8C/S.

- Log

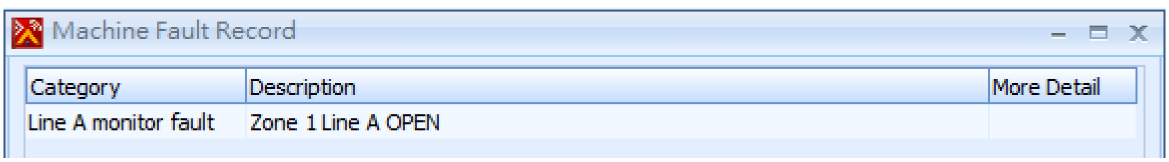
  ○ Line A/B Open

  Text messages display at touch screen on front panel of IDA8C:

  "Line A Error", where Line A can be Line B.

  "Z1 Line A OPEN", where Z1 can be any zone, Line A can be Line B.

  Text message display at window [Machine Fault Record] of Ateis Studio:



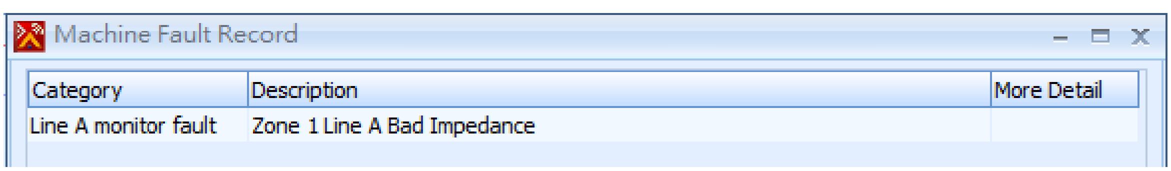
  ○ Line A/B Bad Impedance

  Text messages display at touch screen on front panel of IDA8C:

  "Line A Error", where Line A can be Line B.

  "Z1 Line A Bad Impedance", where Z1 can be any zone, Line A can be Line B.

  Text message display at window [Machine Fault Record] of Ateis Studio:



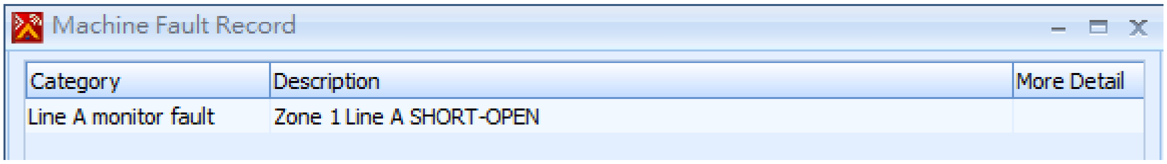
  ○ Line A/B Short

  Text messages display at touch screen on front panel of IDA8C:

  "Line A Error", where Line A can be Line B.

  "Z1 Line A SHORT-OPEN", where Z1 can be any zone, Line A can be Line B.

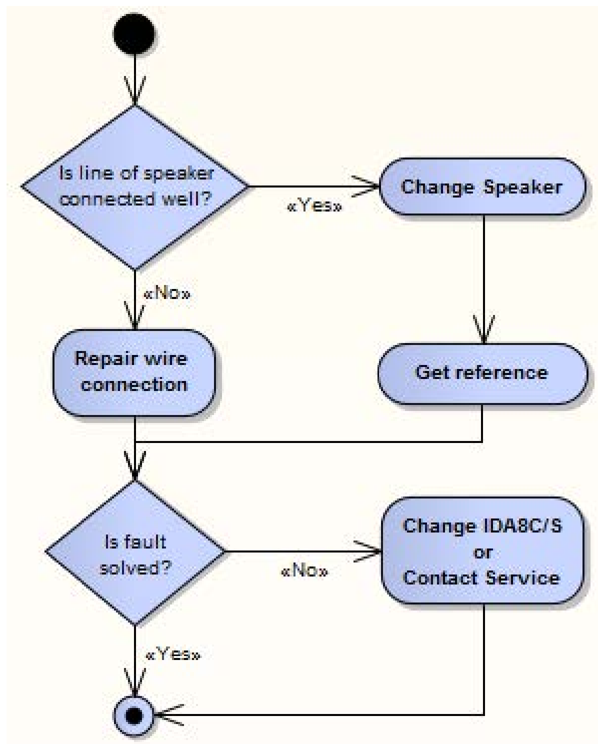
  Text message display at window [Machine Fault Record] of Ateis Studio:


Machine Fault Record

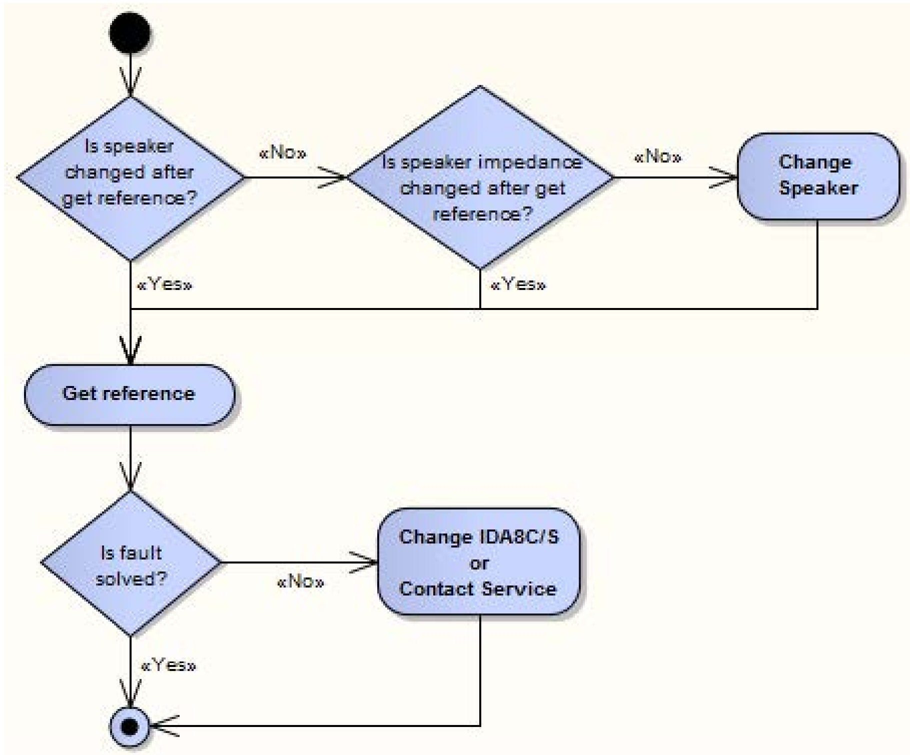
Category	Description	More Detail
Line A monitor fault	Zone 1 Line A SHORT-OPEN	

- Buzzer sound output to monitoring speaker on front panel of IDA8C.

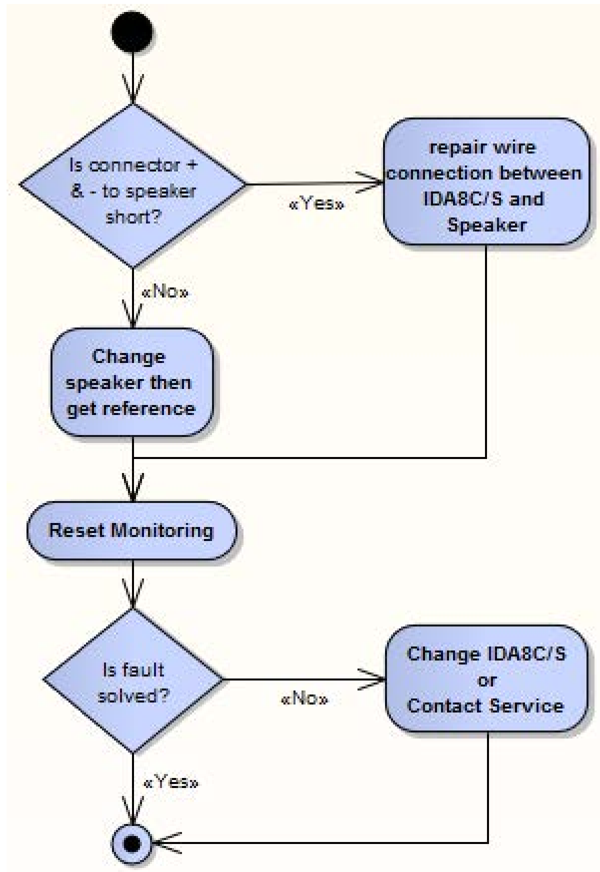
❖ Fault Handling



Line A/B Open



Line A/B Bad Impedance



Line A/B Short

4.1.6.2.3 Amplifier Line Leakage Error

❖ Fault Meaning

- An electricity leakage between IDA8C/S and speaker is detected.
- An electricity leakage between amplifier output and IDA8C/S is detected.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  o G. Fault LED on front panel of IDA8C/S.

  o Zone Fault LEDs on front panel of IDA8C/S.

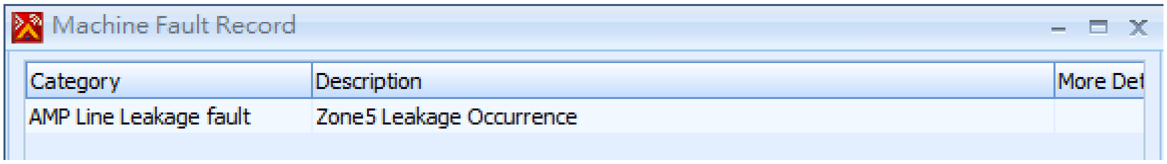
- Log

  Text messages display at touch screen on front panel of IDA8C:

  "AMP Line Leakage Error".

  "Z1 Line Leakage Error", where Z1 can be any zone.

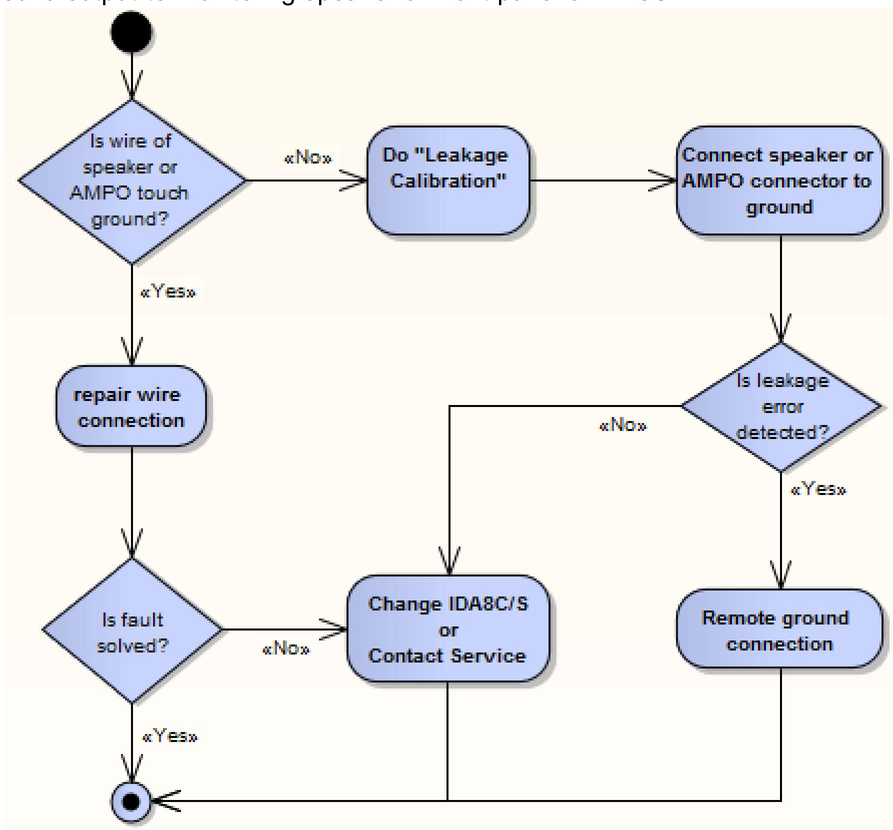
Text message display at window [Machine Fault Record] of Ateis Studio:



Machine Fault Record

Category	Description	More Det
AMP Line Leakage fault	Zone5 Leakage Occurrence	

- Buzzer sound output to monitoring speaker on front panel of IDA8C.



Fault Handling

4.1.6.2.4 Evacuation Intput Error

❖ Fault Meaning

- When evacuation input is set to contact mode, if the voltage of EVAC input is not in the acceptable range.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  - G. Fault LED on front panel of IDA8C/S.

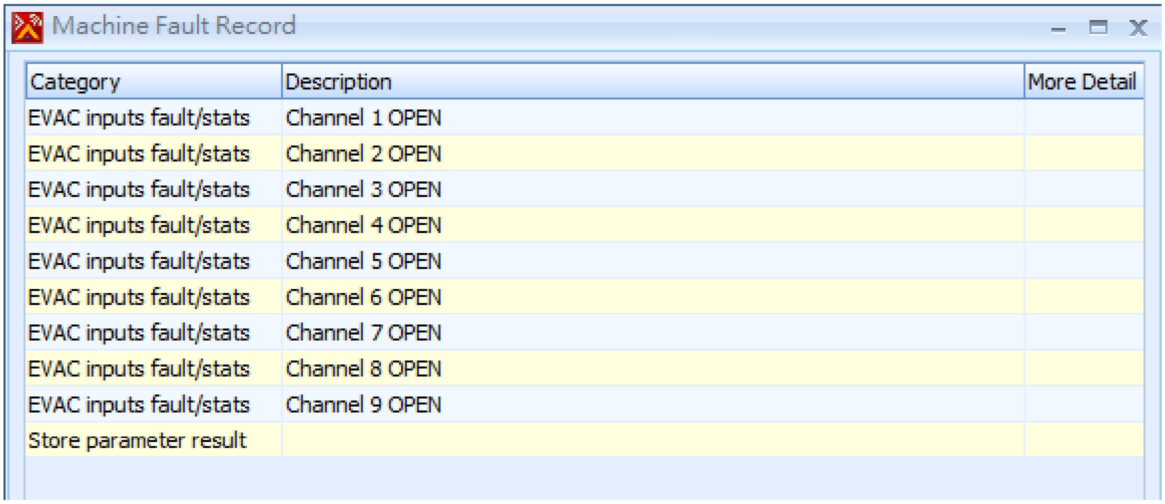
- Log

  - EVAC Input Error(OPEN)

Text messages display at touch screen on front panel of IDA8C:

"EVAC Input Error".

Text message display at window [Machine Fault Record] of Ateis Studio:


Machine Fault Record

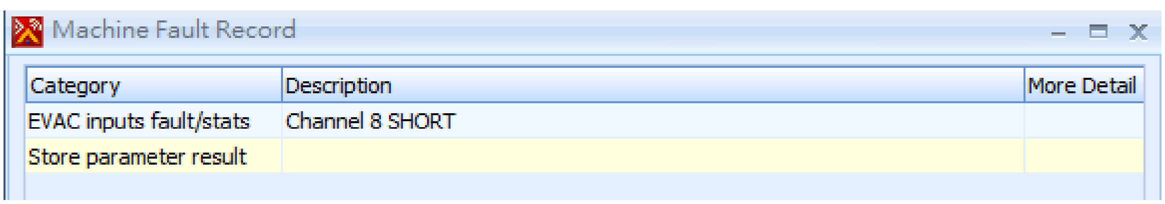
Category	Description	More Detail
EVAC inputs fault/stats	Channel 1 OPEN	
EVAC inputs fault/stats	Channel 2 OPEN	
EVAC inputs fault/stats	Channel 3 OPEN	
EVAC inputs fault/stats	Channel 4 OPEN	
EVAC inputs fault/stats	Channel 5 OPEN	
EVAC inputs fault/stats	Channel 6 OPEN	
EVAC inputs fault/stats	Channel 7 OPEN	
EVAC inputs fault/stats	Channel 8 OPEN	
EVAC inputs fault/stats	Channel 9 OPEN	
Store parameter result		

- EVAC Input Error(SHORT)

Text messages display at touch screen on front panel of IDA8C:

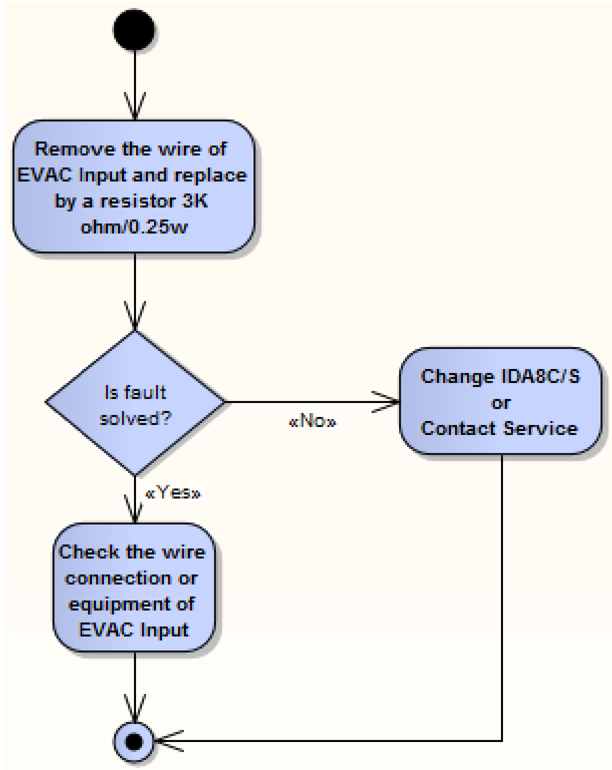
"EVAC Input Error".

Text message display at window [Machine Fault Record] of Ateis Studio:


Machine Fault Record

Category	Description	More Detail
EVAC inputs fault/stats	Channel 8 SHORT	
Store parameter result		

- Buzzer sound output to monitoring speaker on front panel of IDA8C.



Fault Handling

4.1.6.2.5 Ateis Net Broken

❖ Fault Meaning

- Ateis Net is broken, this problem may comes from wire connection between devices on Ateis Net or net card on device.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ G. Fault LED on front panel of IDA8C/S.

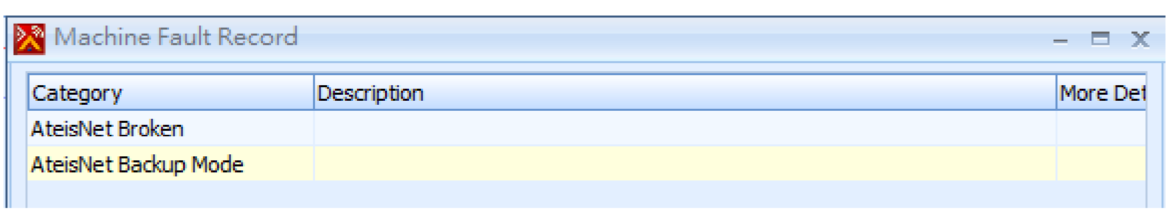
- Log

  Text messages display at touch screen on front panel of IDA8C:

  "AteisNet Broken"

  "AteisNet Backup Mode"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

- Check wire connection between all IDA8C/S on Ateis Net.
- If wire connection is fine, now we can check device:
  - For IDA8C:

    Prepare a good IDA8S, establish Ateis Net wire connection between IDA8C and IDA8S. Deploy Ateis Net using Ateis Studio. If fault is not solved, change Ateis Net card of IDA8C.

  - For each IDA8S on Ateis Net:

    Prepare a good IDA8C, establish Ateis Net wire connection between IDA8C and IDA8S. Deploy Ateis Net using Ateis Studio. If fault is not solved, change Ateis Net card of IDA8S.

4.1.6.2.6 Fireman Microphone Error

❖ Fault Meaning

- Fireman microphone is error.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  - G. Fault LED on front panel of IDA8C/S.

- Log
  - Fireman Microphone Capsule Open

    Text messages display at touch screen on front panel of IDA8C:

      "Fireman Error"

      "Fireman Audio Open"

    Text message display at window [Machine Fault Record] of Ateis Studio:

Machine Fault Record

Category	Description	More Detail
Fireman microphone fault	Fireman Audio Open	

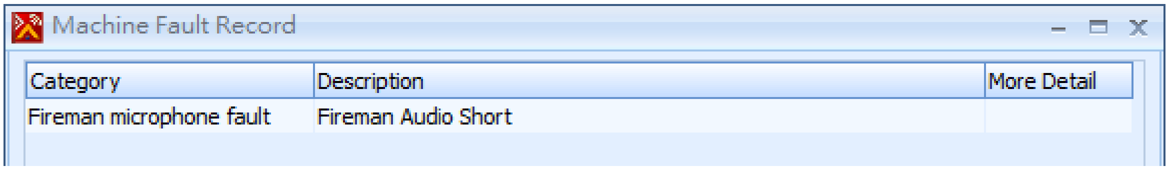
  - Fireman Microphone Capsule Short

    Text messages display at touch screen on front panel of IDA8C:

      "Fireman Error"

      "Fireman Audio Short"

    Text message display at window [Machine Fault Record] of Ateis Studio:



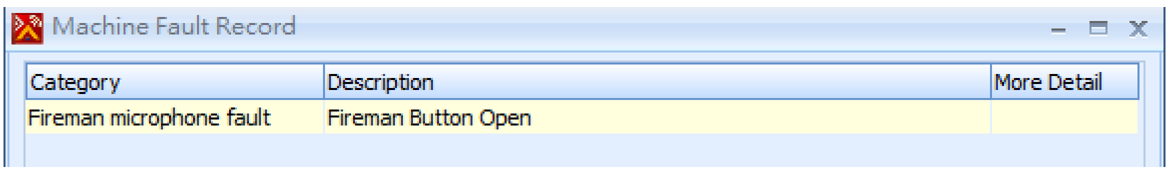
- Fireman Microphone Button Open

  Text messages display at touch screen on front panel of IDA8C:

  "Fireman Error"

  "Fireman Button Open"

  Text message display at window [Machine Fault Record] of Ateis Studio:



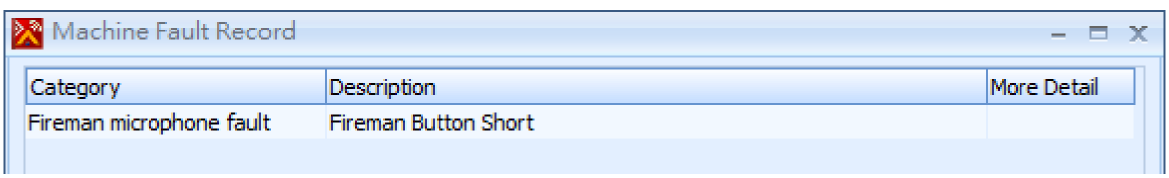
- Fireman Microphone Button Short

  Text messages display at touch screen on front panel of IDA8C:

  "Fireman Error"

  "Fireman Button Short"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

- Check the wire connection of fireman microphone.
- If wire connection is fine but still get fault, change fireman microphone to a good one.
- If fireman microphone is changed but still get fault, change IDA8C or contact service.

4.1.6.2.7 VOX@NET Error

❖ Fault Meaning

- IDA8C/S can't communicate with VOX@NET server.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

- G. Fault LED on front panel of IDA8C/S.

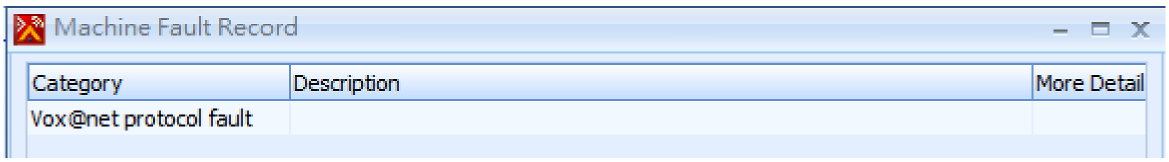
- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Fireman Error"

  "Fireman Audio Open"

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Observe link and active LED indicator on RJ45 connector of PC with VOX@NET server installed and ethernet switch, it is error if link LED is not light up or active LEC isn't blinking. possible cases are listed below:

- Both switch and PC get wrong LED display behavior, check cable between PC & ethernet swich.
- Only VOX@NET get wrong LED display behavior, check network card of PC, you can change a network card to see if it is working.
- Only switch get wrong LED display behavior, check switch, you can simply change to another port to see if it is working.

Observe link and active LED indicator on RJ45 connector of IDA8C/S and switch, possible cases are listed below:

- Both switch and IDA8C/S get wrong LED display behavior, check cable between IDA8C/S & ethernet swich.
- Only IDA8C/S get wrong LED display behavior, check IDA8C/S, you can replace it by a good IDA8C/S to see if it is working.
- Only switch get wrong LED display behavior, check switch, you can simply change to another port to see if it is working.

4.1.6.2.8 User Define Error

❖ Fault Meaning

- A user define error is detected by IDA8C/S, this kind of error can be defined using "Fault Definer" DSP component.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

○ G. Fault LED on front panel of IDA8C/S.

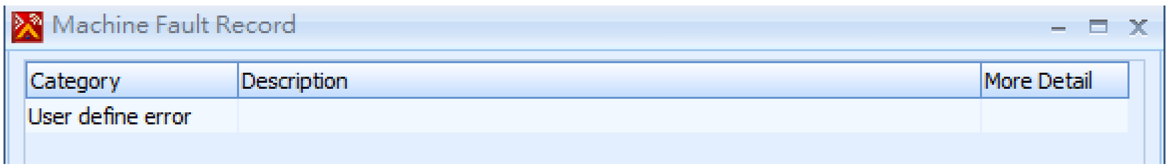
- Log

  Text messages display at touch screen on front panel of IDA8C:

  "User Define Er."

  "XXXXX", where XXXXX is the content defined in "Fault Definer" DSP component field [Message].

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Define on the configuration of DSP component. Basically, you need check the component which is the input of "Fault Definer".

4.1.6.2.9 Remote Offline

❖ Fault Meaning

- IDA8C/S can't communicate with peripheral devices.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ G. Fault LED on front panel of IDA8C/S.

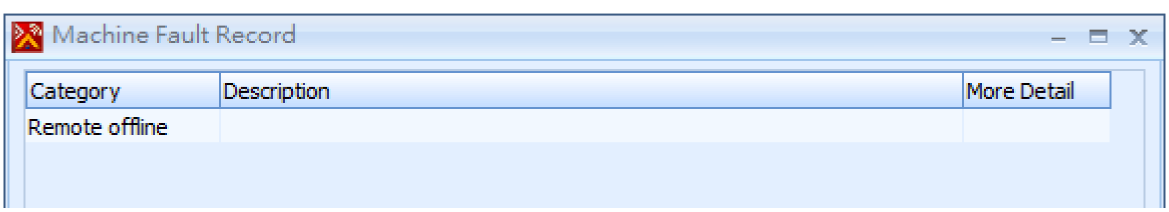
- Log

  Text messages display at touch screen on front panel of IDA8C:

  "Remote Offline"

  "PPM Port:2 ID:1", where Port N is the N^{th} PDC port for peripheral connection, ID is the unique number for each peripheral device on the PDC port.

  Text message display at window [Machine Fault Record] of Ateis Studio:



- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Replace the peripheral device by a good one, if the fault is solved, the problem is coming from peripheral device, send the peripheral device and contact service. If fault is not solved, change IDA8C/S and contact service.

4.1.6.2.10 URGP Fault

❖ Fault Meaning

- Evacuation inputs or fault inputs on URGP are error.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  - G. Fault LED on front panel of IDA8C/S.

- Log

  - EVAC Input Error(Open)

    Text messages display at touch screen on front panel of IDA8C:

    "Evac Board Fault"

    "Port:1 IP:1, ch1 Open", where Port N is the N^{th} PDC port for peripheral connection, IP is the unique number for each peripheral device on the PDC port, ch is the number of channel.

    Text message display at window [Machine Fault Record] of Ateis Studio:

  - EVAC Input Error(Short)

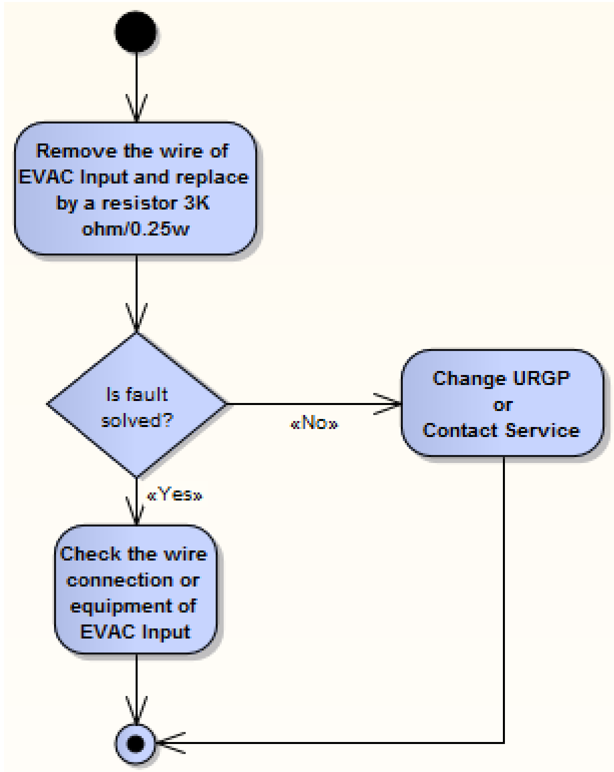
    Text messages display at touch screen on front panel of IDA8C:

    "Evac Board Fault"

    "Port:1 IP:1, ch1 Short", where Port N is the N^{th} PDC port for peripheral connection, IP is the unique number for each peripheral device on the PDC port, ch is the number of channel.

    Text message display at window [Machine Fault Record] of Ateis Studio:

- Buzzer sound output to monitoring speaker on front panel of IDA8C.



Fault Handling

4.1.6.2.11 Remote Fault

❖ Fault Meaning

- An error is detected on peripheral devices. For PPM AS or PSS AS, this fault is about capsule microphone or speaker error.

❖ Fault Indication

- LED Indicator

  When this error is detected, following LEDs will light up:

  ○ G. Fault LED on front panel of IDA8C/S.

- Log

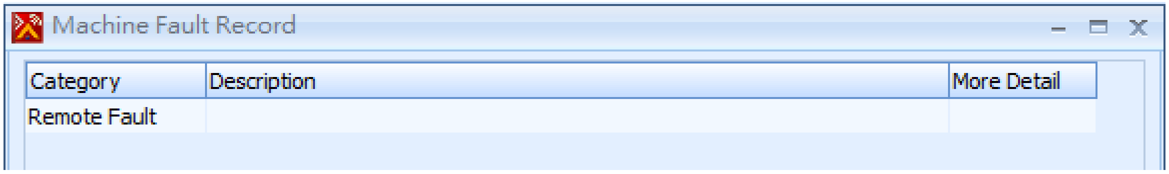
  ○ PPM AS / PSS AS Capsule Microphone Open

    Text messages display at touch screen on front panel of IDA8C:

      "Remote Fault"

      "PSS Port:1 ID:1, Microphone Fault", where Port N is the N^{th} PDC port for peripheral connection, ID is the unique number for each peripheral device on the PDC port.

    Text message display at window [Machine Fault Record] of Ateis Studio:



- PPM AS / PSS AS Capsule Microphone Short

  Text messages display at touch screen on front panel of IDA8C:

  "Remote Fault"

  "PSS Port:1 ID:1, Microphone Fault", where Port N is the Nth PDC port for peripheral connection, ID is the unique number for each peripheral device on the PDC port.

  Text message display at window [Machine Fault Record] of Ateis Studio:

- PPM AS / PSS Speaker Error

  Text messages display at touch screen on front panel of IDA8C:

  "Remote Fault"

  "PSS Port:1 ID:1, Speaker Fault", where Port N is the Nth PDC port for peripheral connection, ID is the unique number for each peripheral device on the PDC port.

  Text message display at window [Machine Fault Record] of Ateis Studio:

- Buzzer sound output to monitoring speaker on front panel of IDA8C.

❖ Fault Handling

Replace the error peripheral device by a good one, and contact service to repair bad one.

4.1.6.3 Procedure of Monitoring Setup

The monitoring setup must be done when hardware setup is appropriate, it means IDA8C/S, amplifiers, speakers, wiring should be prepared and connected before monitoring setup, because IDA8C/S will get values of hardware initial state for further reference, when system is running and reference values are appropriate achieved, IDA8C/S continuity measure internal status and compare with referenced values to determine if there are something wrong of zone/amplifiers or not.

1. Setup hardware equipment, including IDA8C/S, normal amplifier, backup amplifier, speakers, wiring between devices,
2. Make a configuration using Ateis Studio.

3. Store configuration, then online.

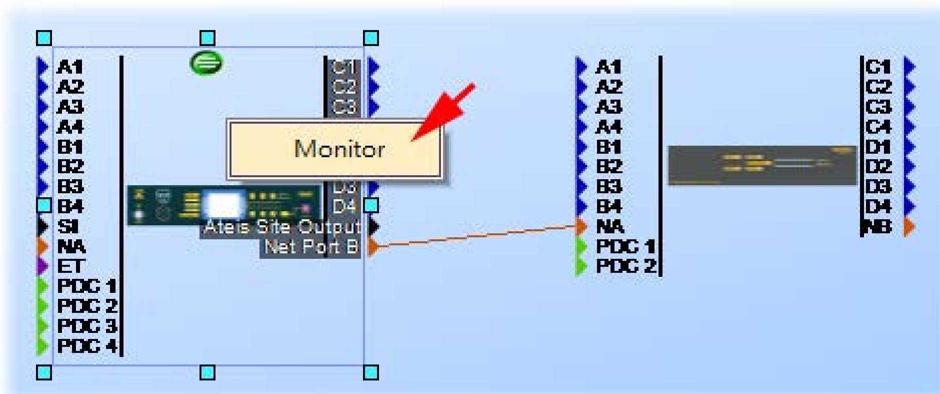
4. Set appropriate settings for your application using Ateis Studio. you can see more details in topic Zone Monitoring, Normal Amplifier Monitoring, Backup Amplifier Monitoring.

5. Press get reference button to start monitoring for the zone you selected.

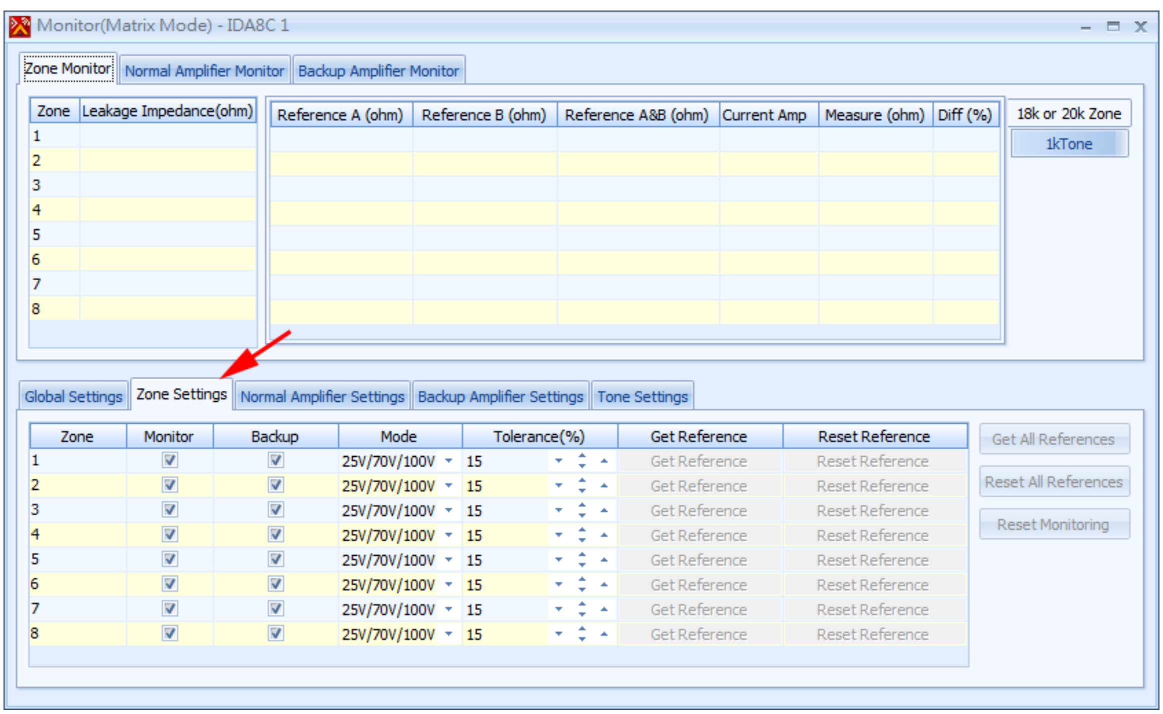
4.1.6.4 Zone Monitoring

❖ Settings

First, in Ateis Studio, right click on IDA8C/S block in [Devices] window:



Window [Monitoring(Matrix Mode)-IDA8C/S X] shows up, select page [Zone Settings]:



There several field in grid of page [Zone Settings] at lower part of window:

- Zone

Indicates a row is refer to which zone.

- Monitor

  Enable/Disable zone monitoring, if monitoring of a zone is disabled, IDA8C/S will not detect faults of zone.

- Backup

  This setting tells IDA8 if backup amplifier can service the zone or not. i.e. if the checkbox is not checked, when a fault of normal amplifier IDA8C/S will not switch to backup amplifier for the zone, result to the monitoring of the zone is not working.

- Mode

  There are two choices:

  1. 25V/70V/100V

     Audio signal go through amplifier and back to IDA8C/S.

  2. 0dB

     This mode the audio is output to other equipment, not go through amplifier and back to IDA8C/S. Monitoring is disabled under this mode.

- Tolerance(%)

  Specify the tolerance of impedance to detect an bad impedance fault. for example if the reference is 1000 ohm, tolerance set to 15%, then the range of good impedance will be 850 ~ 1150 ohm, if measured impedance value is 1300 ohm, a bad impedance fault is recognized by IDA8C/S.

- Get Reference

  When system is construction, user need to get the impedance value of speaker for further reference, this value will utilized to detect fault of zones. The rule of fault detection is described in preview item "Tolerance(%)".

- Reset Reference

  Reset the reference value to empty, result to disable zone monitoring. but the measure value still updated.

There are buttons in the right side of page [Zone Settings]:

- Get All Reference

  Get reference for all zones.

- Reset All Reference

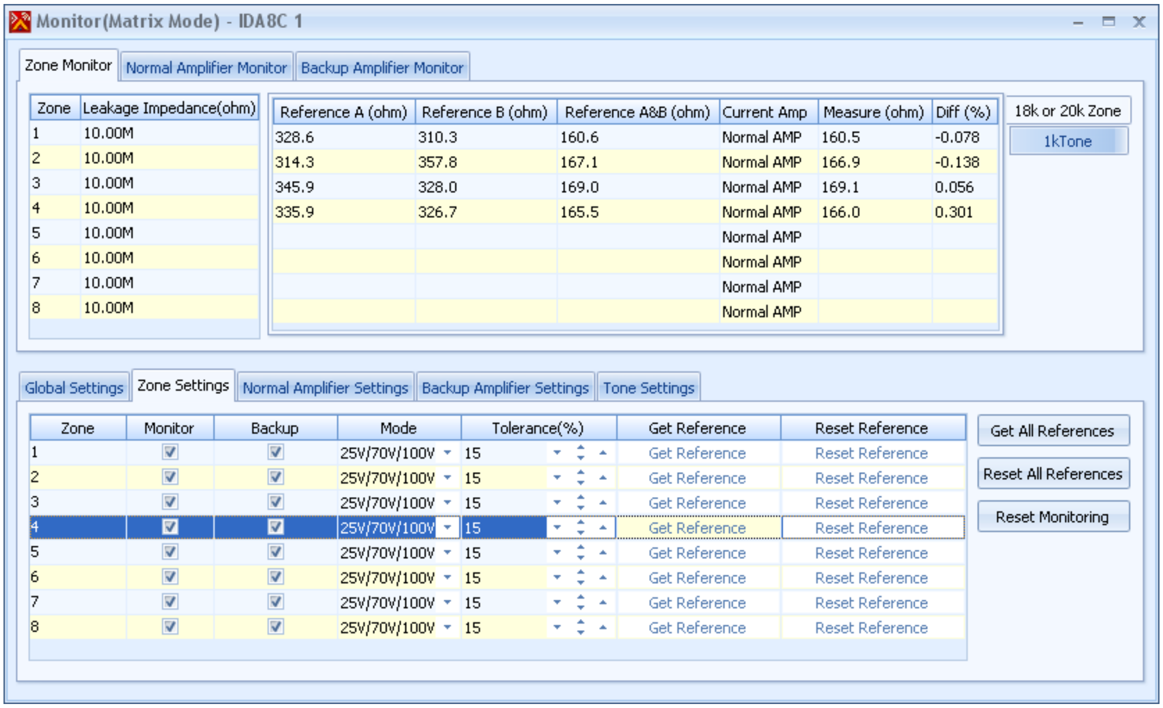
  Reset reference for all zones.

- Reset Monitoring

  This will tell IDA8C/S to reset monitoring, to initial state of device. In the case of zone short fault, IDA8C/S will open the internal relay to stop output audio signal to speaker. after the relay is opened, IDA8C/S has no way to know if the circuit outside is short or not, so after user repair the circuit, it needs to reset monitoring to switch relay close to output audio signal to speaker again, then IDA8C/S can monitor the short problem again.

❖ Measurement

After parameter settings are done in lower part grid, IDA8C/S start to measure value get from internal circuit, and display it on the grid at upper part of window:



There are fields of the grid:

- Leakage Impedance

  The impedance between LineA/B +/- signal and ground cause current to leak.

- Reference A (ohm)

  Show the impedance value for reference get from Line A output.

- Reference B (ohm)

  Show the impedance value for reference get from Line B output.

- Reference A&B (ohm)

  Show the impedance value for reference get from Line A & B.

- Current Amp

  Show the amplifier for the zone is normal amplifier or backup amplifier.

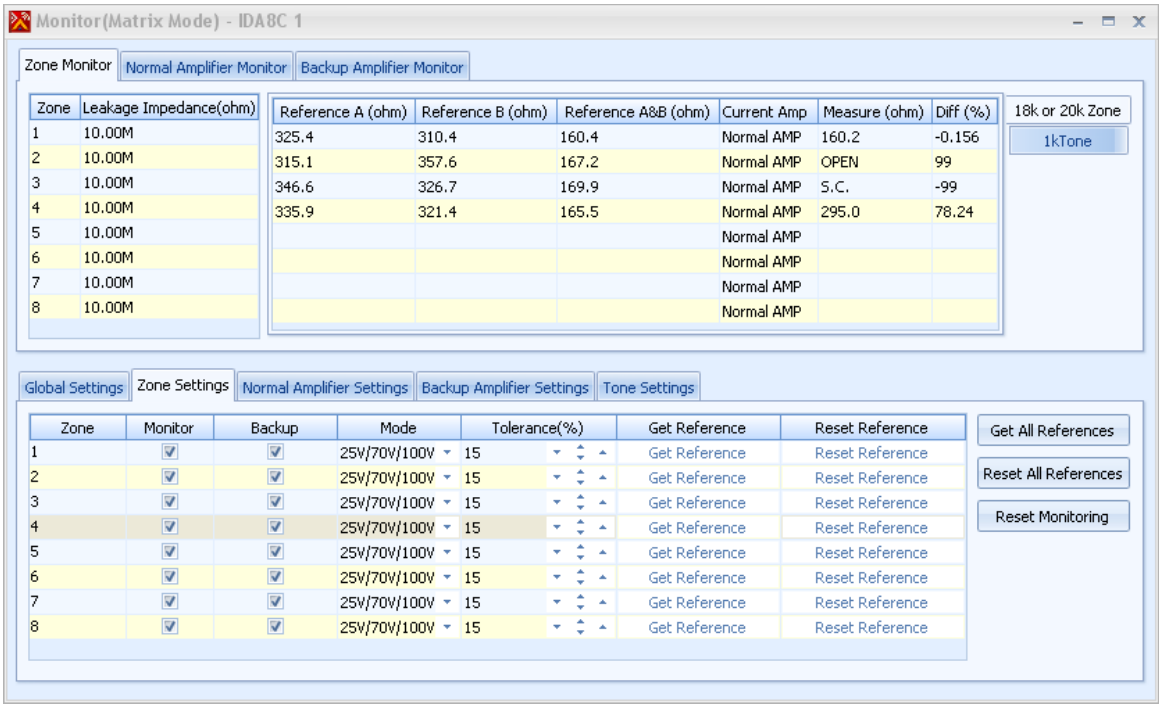
- Measure (ohm)

  Show the impedance value of Line A & B.

- Diff(%)

  Show the difference percentage of Measure(ohm) and Reference A&B(ohm)

❖ Faults



Leakage occurrence:

If leakage impedance lower than the leakage threshold which is defined in [Global Settings] tab, then a Leakage Occurrence fault is recognized.

If the internal values of monitoring Line A/B become not normal, IDA8C/S will detect and recognize that error, then generate a global fault Line A/B Error, the error status is showed in the field [Measure(ohm)]. There are three kinds of Line A/B Error listed below:

- Line A/B Open

  If the value of Measure (ohm) is greater than 4000 ohm, this fault can be recognized by IDA8C/S, In the field of [Measure(ohm)] shows "OPEN".

- Line A/B Bad Impedance

  If the value of Measure (ohm) is greater than Reference A&B (ohm) + Tolerance (%), or value of Measure (ohm) is less than Reference A&B (ohm) - Tolerance (%), this fault can be recognized by IDA8C/S. In the field of [Diff(%)] shows the percentage of difference, in above figure, the difference percentage of zone 4 is 78.24 which is great than the tolerance 15%, a Line A/B bad impedance is recognized.

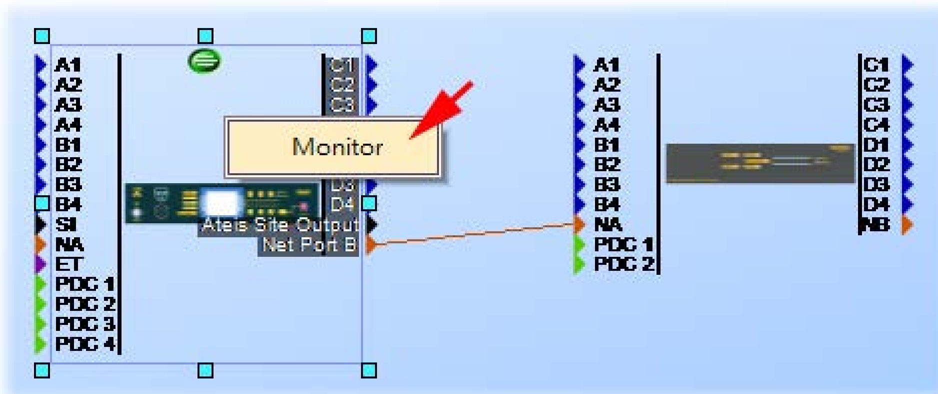
- Line A/B Short

  The voltage of pin +, - is monitored by IDA8C/S, if the voltage is zero, it means there is a short of circuit outside IDA8C/S, if this fault is detected, the internal relay of zone will be open to stop audio signal output to speaker. User need to do a Reset Monitoring described in preview part of this topic to switch the status back, i.e. make audio output to speaker and enable monitoring of short again. A string "S.C." is showed in field [Measure(ohm)]

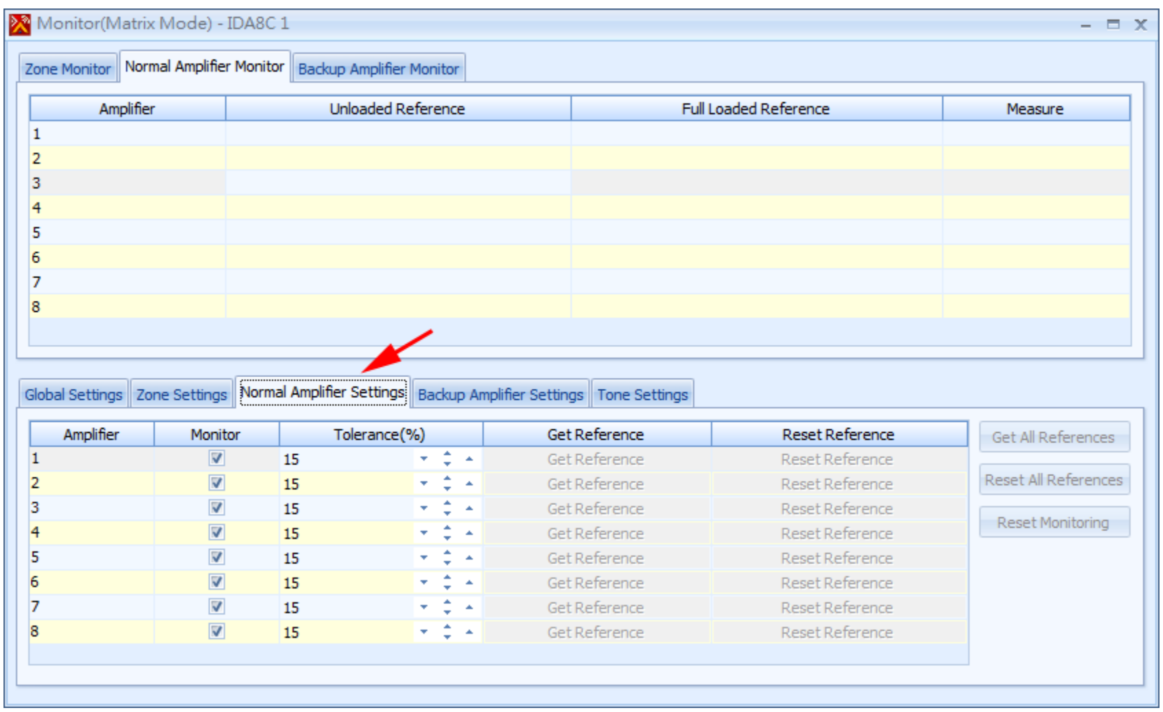
4.1.6.5 Normal Amplifier Monitoring

❖ Settings

First, in Ateis Studio, right click on IDA8C/S block in [Devices] window:



Window [Monitoring(Matrix Mode)-IDA8C/S X] shows up, select page [Normal Amplifier Settings]:



- Amplifier

  Indicates a row is refer to which normal amplifier of zone.

- Monitor

  Enable/Disable normal amplifier monitoring for selected zone, if monitoring of a normal amplifier is disabled, IDA8C/S will not detect faults of normal amplifier.

- Tolerance(%)

To determine the boundary of fault detection for selected zone. There is a fault recognized if one of follow conditions is true:

- Measure < (Full Loaded Reference x (100% - Tolerance%))
- Measure > (Unloaded Reference x (100% + Tolerance%))

- Get Reference

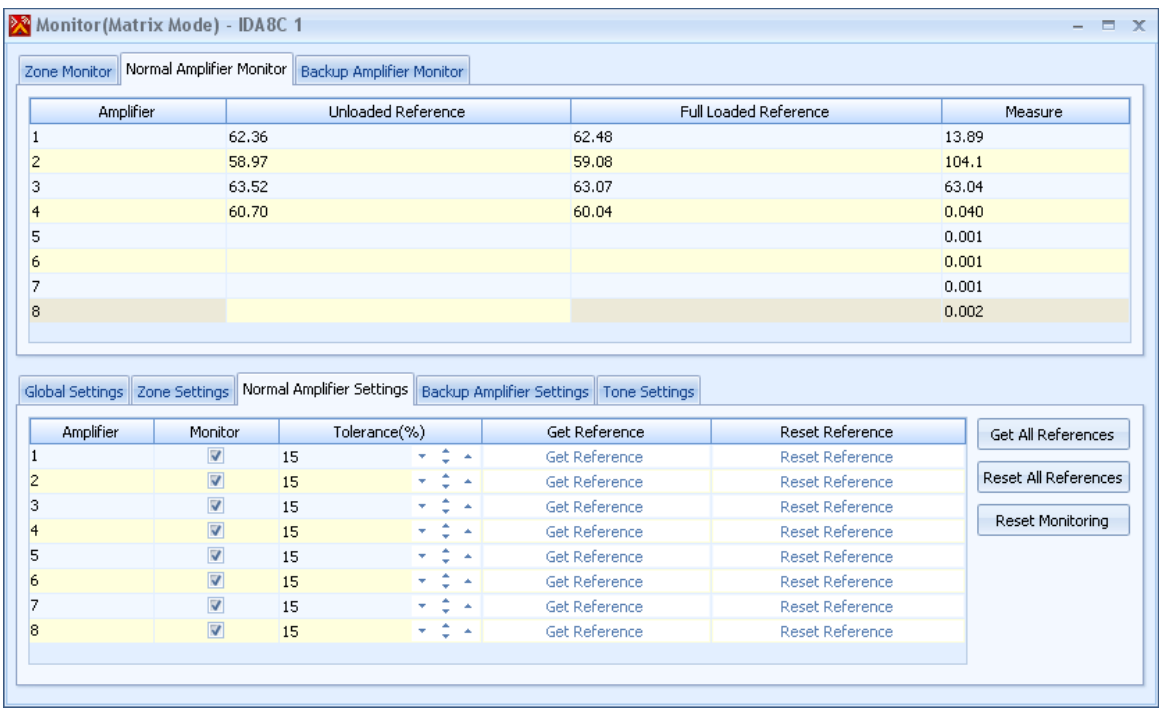
  Get the reference values of selected zone for further fault detection. Be aware, hardware configuration(IDA8C/S, amplifiers, wiring) should be done before get reference.

- Reset Reference

  Clear reference values of selected zone to go back initial status. if reference value is cleared, monitoring will be stopped.

❖ Measurement

After parameter settings are done in lower part grid, IDA8C/S start to measure value get from internal circuit, and display it on the grid at upper part of window:



- Unloaded Reference

  Display the value of unloaded reference which is the gain of amplifier when there is no load to the zone. This value use to determine if there is a fault of normal amplifier or not.

- Full Loaded Reference

  Display the value of full loaded reference which is the gain of amplifier when there is full load to the zone. This value use to determine if there is a fault of normal amplifier or not.

- Measure

  Display the value of measured which is the gain of amplifier.

❖ Faults

- Amplifier Gain Too High

  If Measure > (Unloaded Reference x (100% + Tolerance%)), an Amplifier Gain Too High fault is recognized. once this fault is happen, do the following checks:

  o Make sure there is good connection between speaker and IDA8C/S.

  o Ensure the speaker is a good one.

  o Check gain of amplifier, maybe the value is changed after get reference.

  If all check passed but still get a amplifier too high fault, change amplifier.

- Amplifier Gain Too Low

  If Measure < (Full Unloaded Reference x (100% - Tolerance%)), an Amplifier Gain Too Low fault is recognized. once this fault is happen, do the following checks:

  o Check if the signal goes into amplifier is too large, some amplifier has an indicator "Overload", you can check this indicator.

  o Check if speaker is good or damaged.

  If all check passed but still get a amplifier too low fault, change amplifier.

  If this fault generated after speaker configuration changed, you can do a get reference again to get new values for reference.

- Amplifier Fail

  Measure < (Unloaded Reference / 10), once this fault is happen, do the following checks:

  o Check if amplifier is power on or not.

  o Check if the gain of amplifier is set to 0, if it is, set to a suitable value.

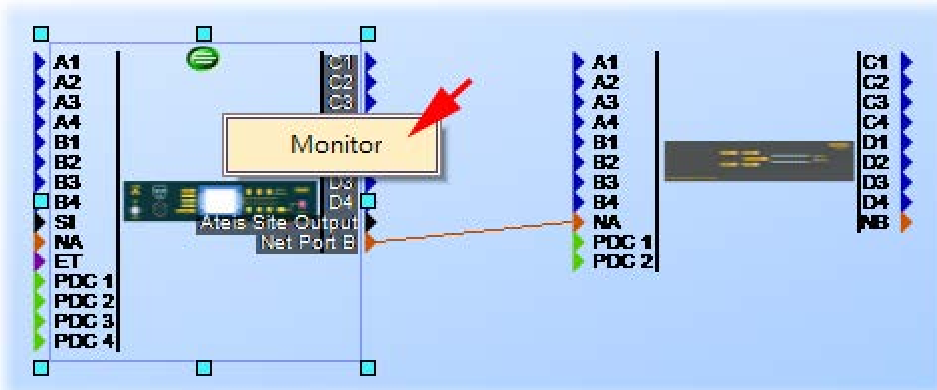
  o Check if the wires between amplifier and IDA8 are good connection.

  If all check passed but still get a amplifier fail fault, change amplifier.

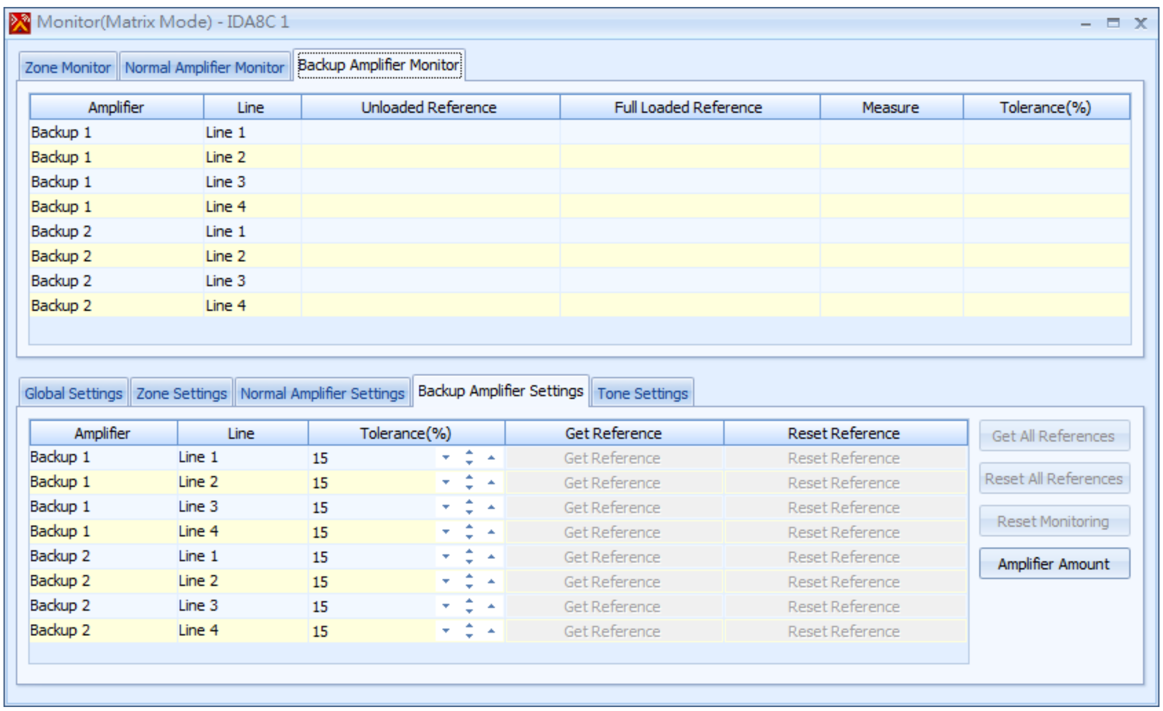
4.1.6.6 Backup Amplifier Monitoring

❖ Settings

First, in Ateis Studio, right click on IDA8C/S block in [Devices] window:



Windows [Monitor(Matrix Mode) - IDA8C/S X] opened, select page [Backup Amplifier Settings]:



- Amplifier

  Indicate which backup amplifier will be monitored.

- Line

  Indicate which zone and backup amplifier combination will be monitored.

- Tolerance(%)

  To determine the boundary of fault detection for selected zone. There is a fault recognized if one of follow conditions is true:

  - Measure < (Full Loaded Reference x (100% - Tolerance%))
  - Measure > (Unloaded Reference x (100% + Tolerance%))

- Get Reference

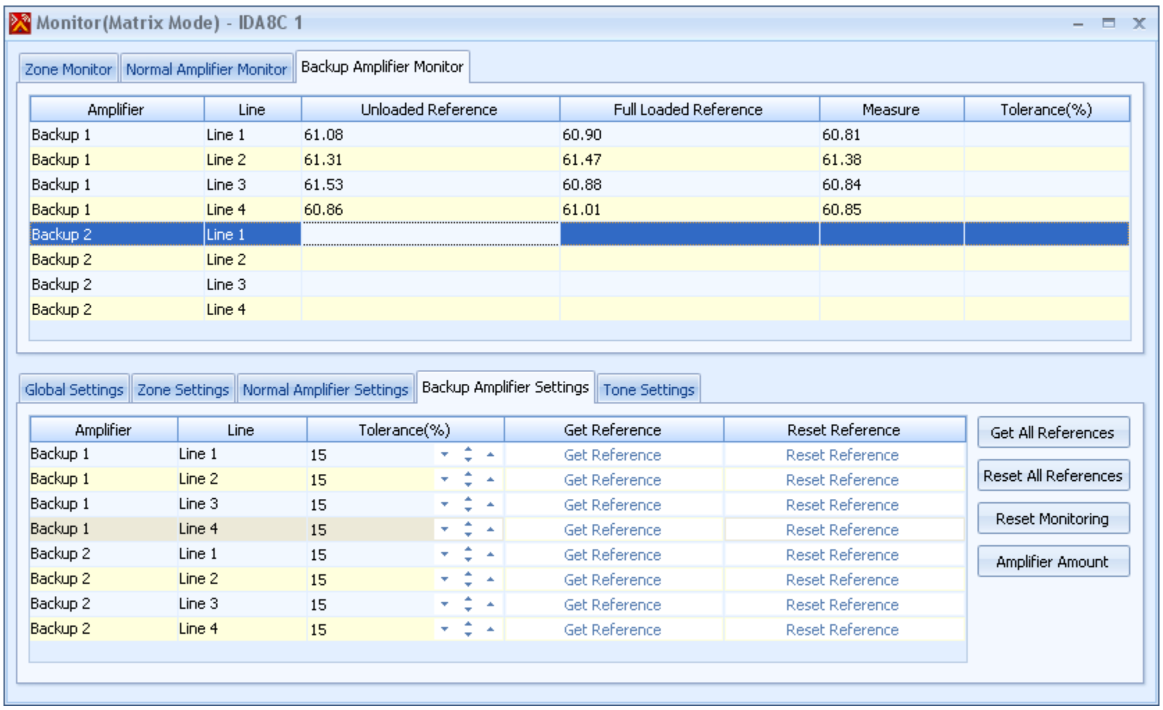
  Get the reference values of selected zone for further fault detection. Be aware, hardware configuration(IDA8C/S, amplifiers, wiring) should be done before get reference.

- Reset Reference

  Clear reference values of selected zone to go back initial status. if reference value is cleared, monitoring will be stopped.

❖ Measurement

After parameter settings are done in lower part grid, IDA8C/S start to measure value get from internal circuit, and display it on the grid at upper part of window:



- Unloaded Reference

  Display the value of unloaded reference which is the gain of amplifier when there is no load to the zone. This value use to determine if there is a fault of normal amplifier or not.

- Full Loaded Reference

  Display the value of full loaded reference which is the gain of amplifier when there is full load to the zone. This value use to determine if there is a fault of normal amplifier or not.

- Measure

  Display the value of measured which is the gain of amplifier.

❖ Faults

- Amplifier Too High

  If Measure > (Unloaded Reference x (100% + Tolerance%)), an Amplifier Too High fault is recognized. once this fault is happen, do the following checks:

- Make sure there is good connection between speaker and IDA8C/S.
- Ensure the speaker is a good one.
- Check gain of amplifier, maybe the value is changed after get reference.

If all check passed but still get a amplifier too high fault, change amplifier.

- Amplifier Too Low

  If Measure < (Full Unloaded Reference x (100% - Tolerance%)), an Amplifier Too Low fault is recognized. once this fault is happen, do the following checks:

  - Check if the signal goes into amplifier is too large, some amplifier has an indicator "Overload", you can check this indicator.
  - Check if speaker is good or damaged.

  If all check passed but still get a amplifier too low fault, change amplifier.

  If this fault generated after speaker configuration changed, you can do a get reference again to get new values for reference.

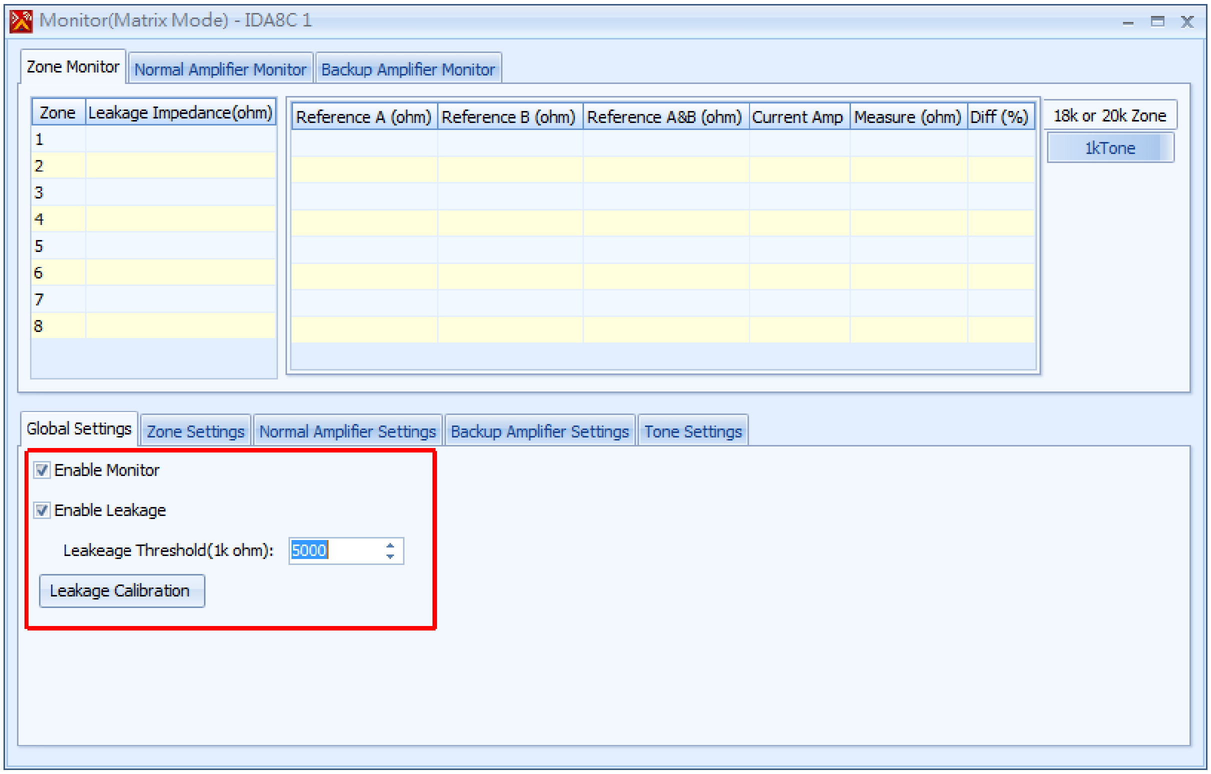
- Amplifier Fail

  Measure < (Unloaded Reference / 10), once this fault is happen, do the following checks:

  - Check if amplifier is power on or not.
  - Check if the gain of amplifier is set to 0, if it is, set to a suitable value.
  - Check if the wires between amplifier and IDA8 are good connection.

  If all check passed but still get a amplifier fail fault, change amplifier.

4.1.6.7 Global Settings



There are global settings listed below:

- Enable Monitor

  A check box to determine enable or disable monitor function.

- Enable Leakage

  A check box to determine enable or disable leakage detect function.

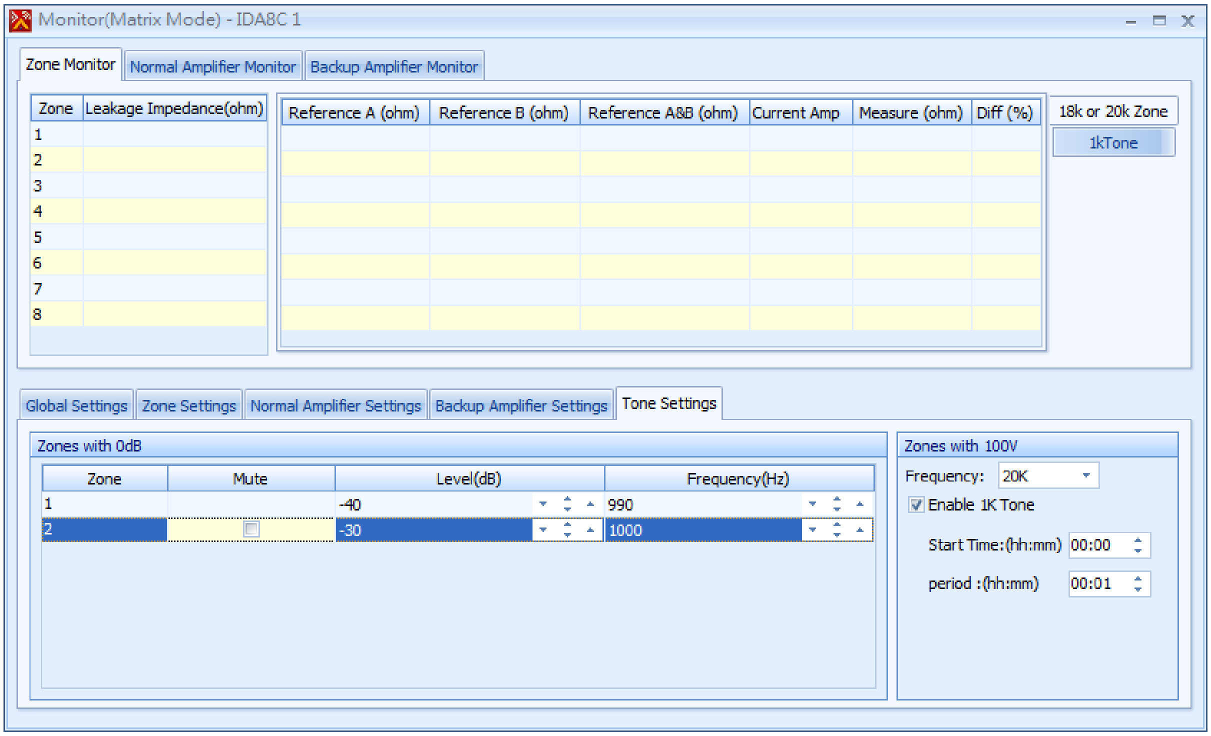
- Leakage Threshold(1k ohm)

  A threshold value for leakage detection. If leakage impedance lower than this value, the fault Leakage Occurrence is detected.

- Leakage Calibration

  If system detect a leakage fault, but the circuit does really not have this problem after doing some propriety check, it need to do a leakage calibration to make monitor function working fine.

4.1.6.8 Tone Settings



In the lower part of window [Monitoring(Matrix Mode)-IDA8C/S X], There is a page [Tone Settings] for configuring tone signal which is utilized for monitoring. There are two kind of settings:

❖ Zones with 0dB

If audio signal from Amplifier Zone Output is not go back IDA8C/S for monitoring, It allow user to mix a tone into that signal for further detection. You can change this setting in [Zone Settings] tab and [Mode] field. There are parameters of settings:

- Zone

  Indicate zone number of parameters.

- Mute

  To mute tone signal.

- Level(dB)

  The level of tone signal.

- Frequency(Hz)

  The frequency of tone signal.

❖ Zones with 100V

- Frequency

  The frequency of tone signal.

- Enable 1K Tone

Enable 1K tone for monitoring. If this option is on, during Start Time and End Time, a 1K tone signal send to amplifier for monitoring.

o Start Time:(hh:mm)

The time to start sending 1K tone.

o End Time:(hh:mm)

The time of end 1K tone sending.

4.1.7 Bypass Mode Paging

IDA8C/SX provide a safe mechanism for paging when microchip inside the device is not working. We call it bypass mode paging. When system is in bypass mode paging, paging source through normal amplifier directly without DSP component processing, and backup amplifier can't working due to micro-controller is crash.

❖ Source of bypass mode paging

There are two source can be the input of bypass mode paging:

- Fireman microphone

  Fireman microphone is a microphone with a button can be plugged in front panel of IDA8. When system under bypass mode paging, press the button to start paging, release button to end paging.

- Audio coming from PDC1

  IDA8C allow you to use PDC1 be the source of bypass paging, it need a button to start paging also, the by-pass mode input contact can connect to a button for triggering bypass mode paging.

⚠ Fireman microphone with higher priority than PDC1 source.

❖ Condition of Bypass Mode Switching

There are two ways to switch system state to bypass mode paging:

- Hardware monitoring

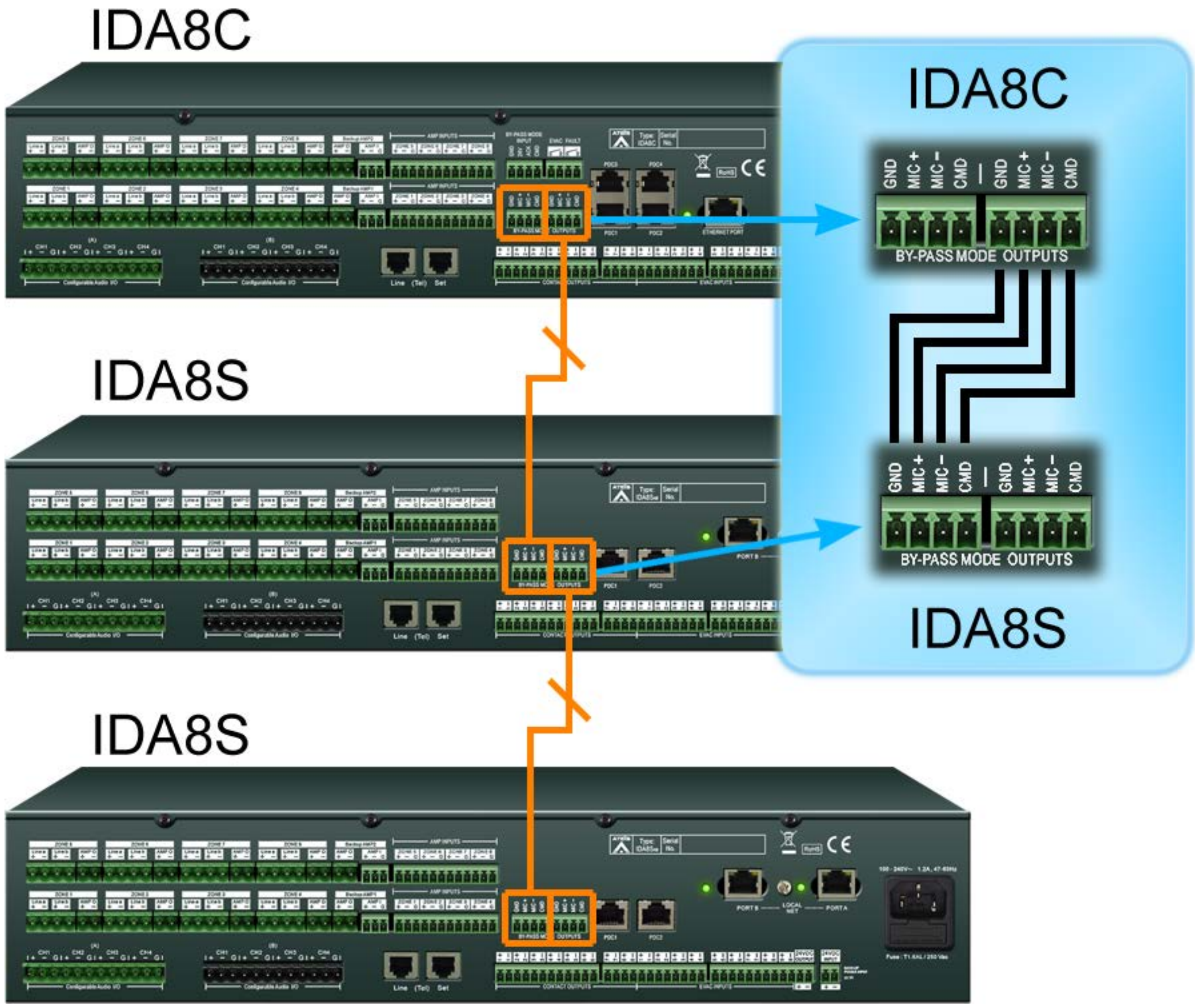
  In IDA8 devices, there is a hardware monitored mechanism to detect micro controller crash, system switch to bypass mode paging if micro controller crash detected.

- Bypass mode input contact

  Connect pin CMD and GND of bypass-mode input to a button, this input contact can trigger bypass mode paging.

❖ cascade

If the application use multiple IDA8s and use fiber optic as Ateis Net transmitting media, you need to do an extra wiring for bypass mode paging cascade, see following figure:

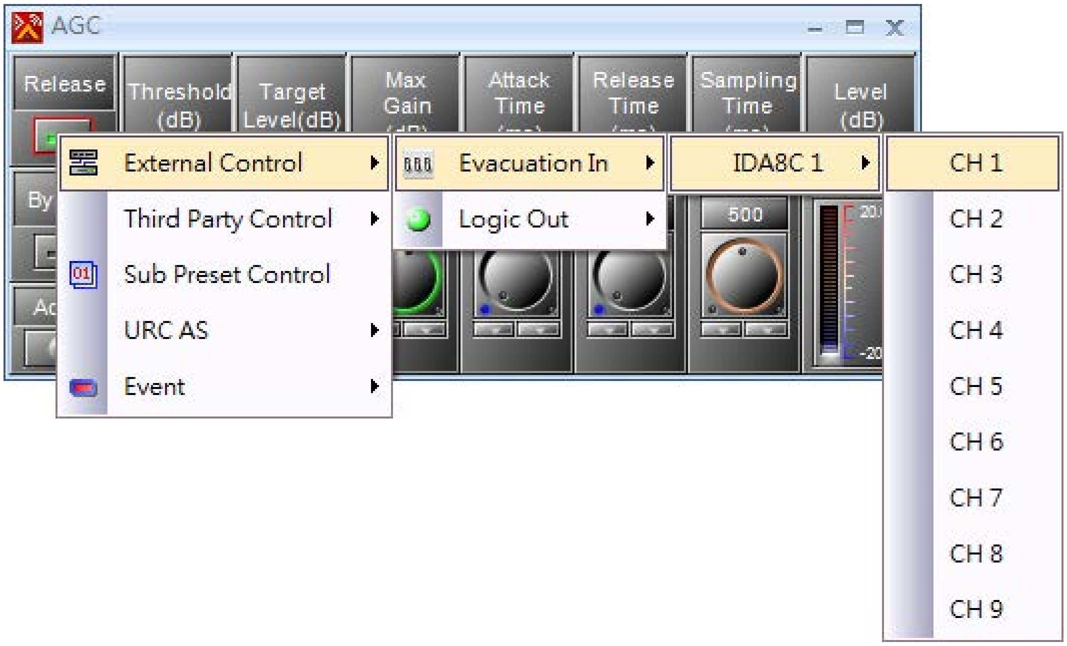


4.1.8 Contact I/O

4.1.8.1 Evacuation Input

IDA8 Devices have evacuation inputs that allow the monitoring of external contact. To config evacuation input, you have two ways to do:

- Assign element to evacuation input control



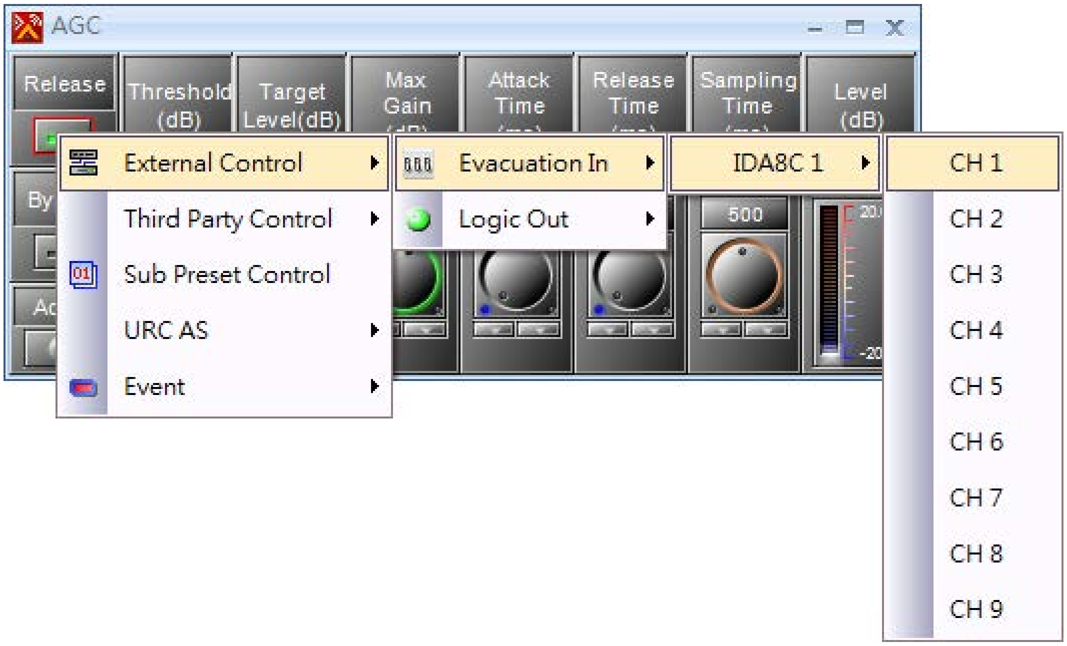
- Use Evacuation Input component
- By use Evacuation Input component, you can control the other logic components. Logic signal is coming from Evacuation Input component and goes any other input pins of logic component as you want. please refer to topic Evacuation Input component.

They also can be used in UGA mode, trigger by a voltage polarization change. To setting this option you need use Evacuation Input component, please refer to topic Evacuation Input component.

4.1.8.2 Contact Output

In the IDA8C/S devices, there are logic outputs channels to close/open circuit for an external device. There are two ways to config it:

- Assign element to evacuation output control



- Use Logic Out component

By using Logic Out component, you can output signal coming from any logic component. please refer to topic Logic Out component.

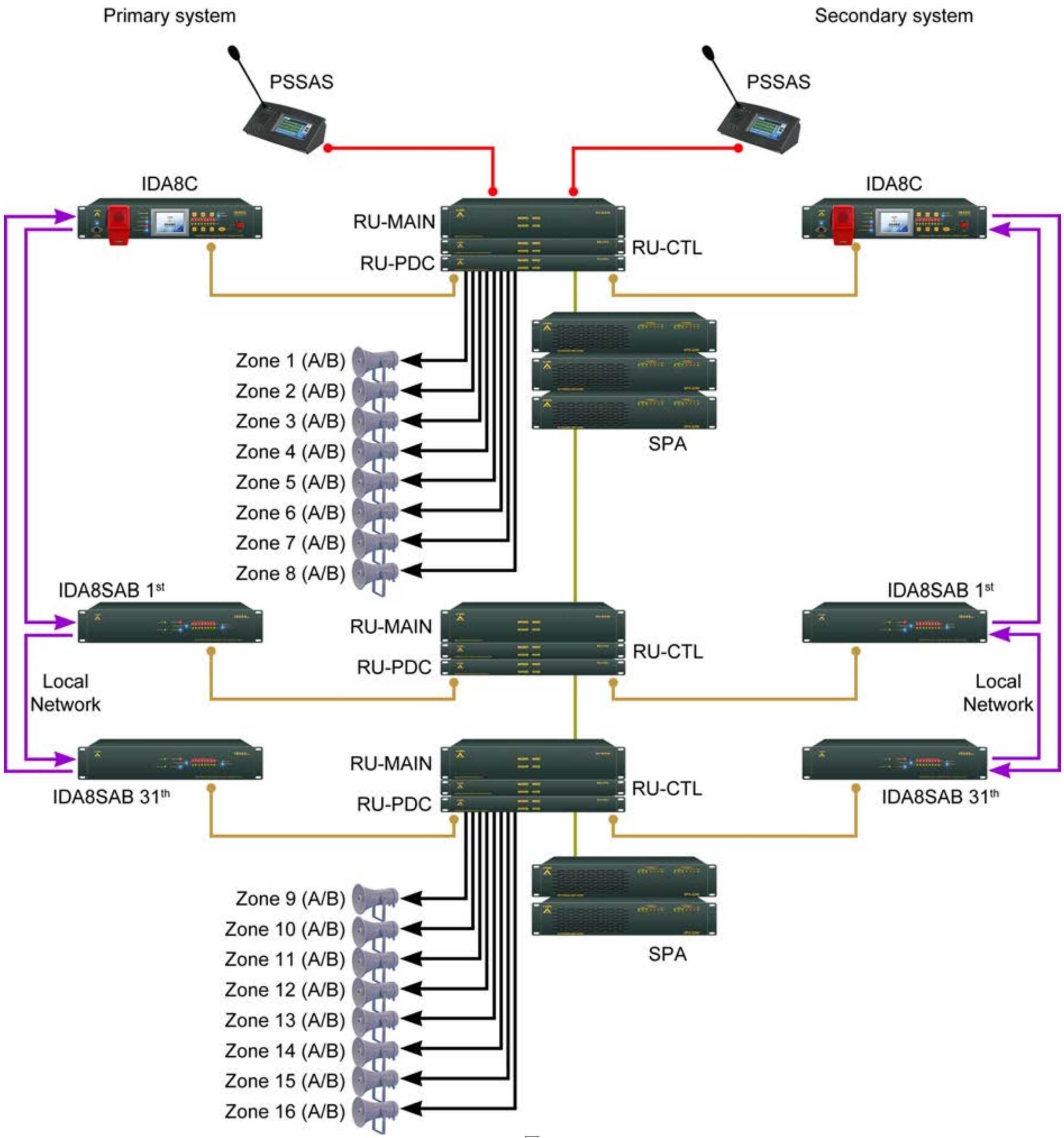
4.2 Consoles and Accessories

4.2.1 Redundancy Unit

4.2.1.1 Overview

PAVA systems in very highly sensitive places like nuclear power centrals, underground industrial systems or places where people have to thrust on a 100% availability of a PAVA system, may require for redundancy on any part of the PAVA system. Redundancy is a very widely-spread expression that needs to be further specified into a required level of redundancy. In normal PAVA systems where other communication means are available, spare amplifiers and surveillance of essential components is more than enough. Higher levels of redundancy may require for A/B wiring of the loudspeaker lines where loss of the A or B line or system still ensures a minimum coverage of 50% of the venue. (Such requirement is detailed described in the BSI-5839 -part 8). But even that could not be enough in situations where the PAVA is the only and last means of communication. In this case an even higher level of redundancy may be required. At this level, not only the amplifiers have redundancy by means of active spare amplifiers, also the central equipment will have a full back-up.

This is what we call: Full-Redundancy. Ateïs RU provides this high level of redundancy. RU is a thorough switching device that acts as a Primary / Secondary switching device



There are three types of redundancy unit:

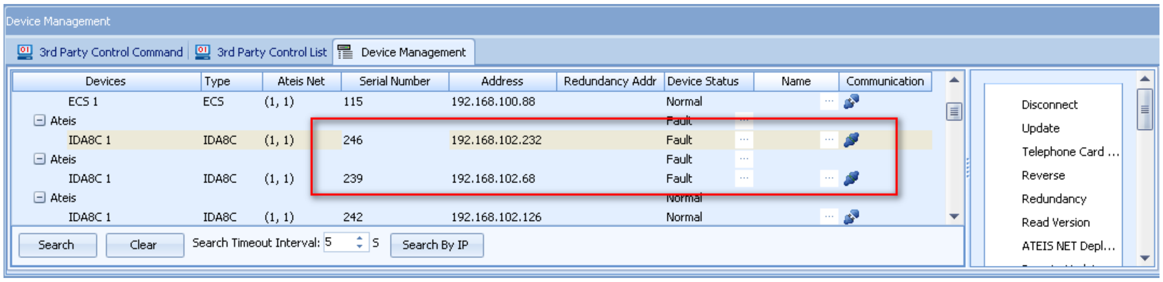
- RU-MAIN.: Switch digital audio and processing matrix like IDA8C, paging consoles and interfaces.
- RU -CTL: Switch security and normal contacts IN and OUT.
- RU -PDC: Switch auxiliary audio IN and OUT , two additional paging consoles and telephone line.

4.2.1.2 Configuration

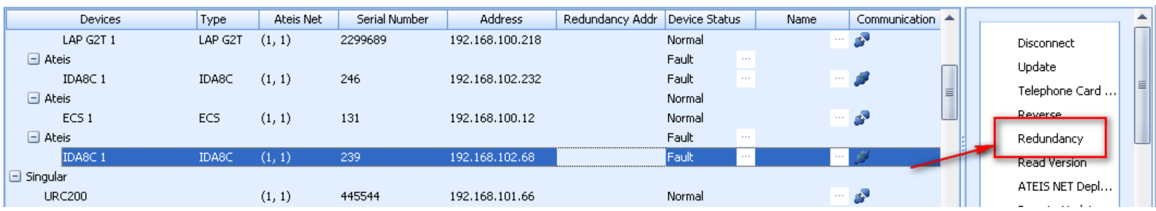
The steps to configuration redundancy system in IDA8 system listed below:

1. Setup H/W connection redundancy system.

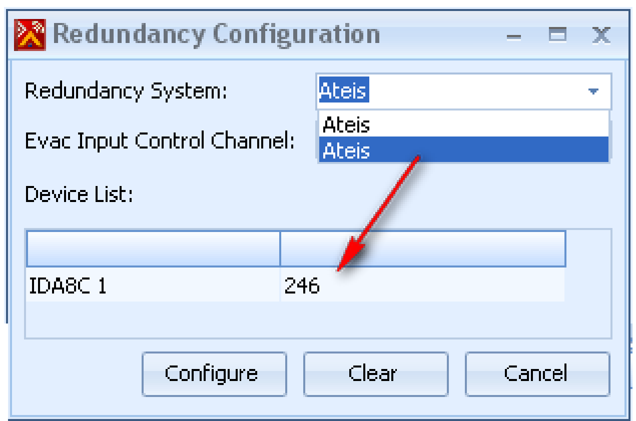
2. Search primary and secondary IDA8C using Ateis Studio, and connect to them.



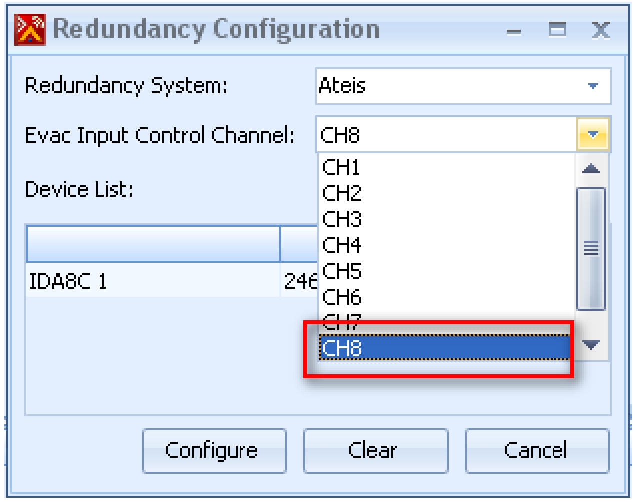
3. Select the IDA8C to be primary, Click button [Rendundancy] on functions window.



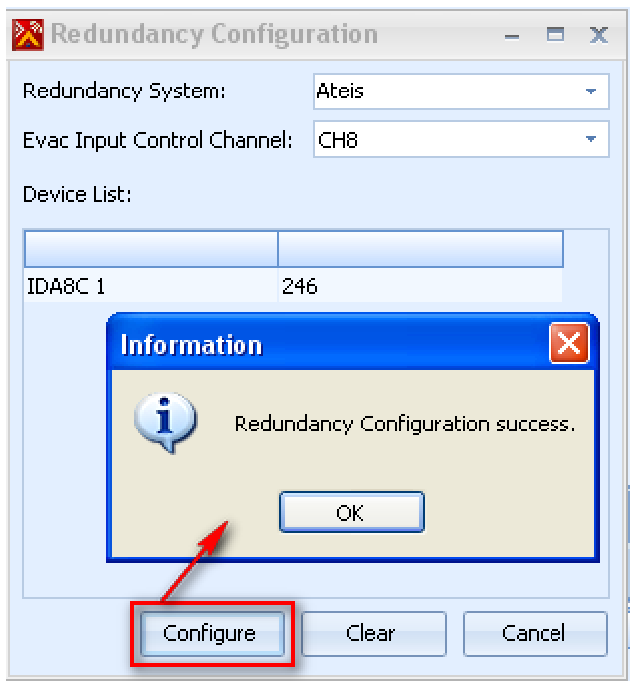
4. Choose the system of secondary IDA8C belongs to. You can click item in combo box of redundancy then check if the device in [device list] grid is desired by it's serial number.

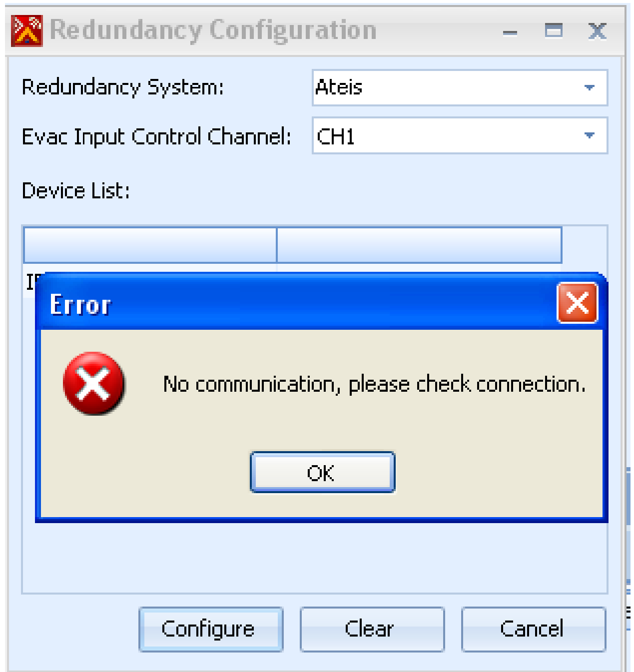


5. Set Evacuation input Control Channel.

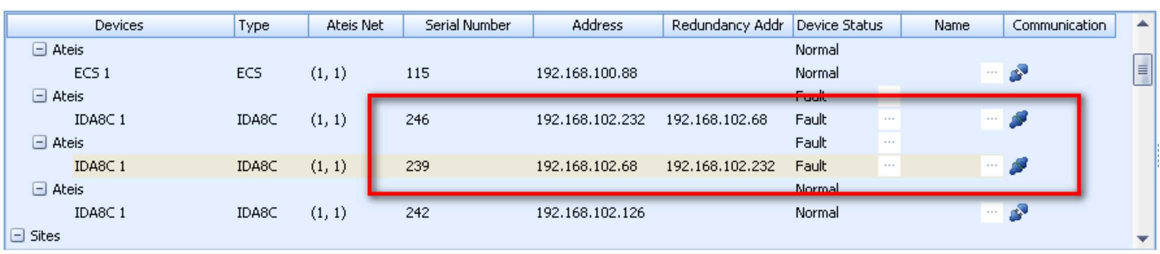


6. Press button [Configure], A window shows message "Redundancy Configuration Success" if the procedure successfully done. Otherwise a error window shows. It the procedure was not success, try to choose the other system.



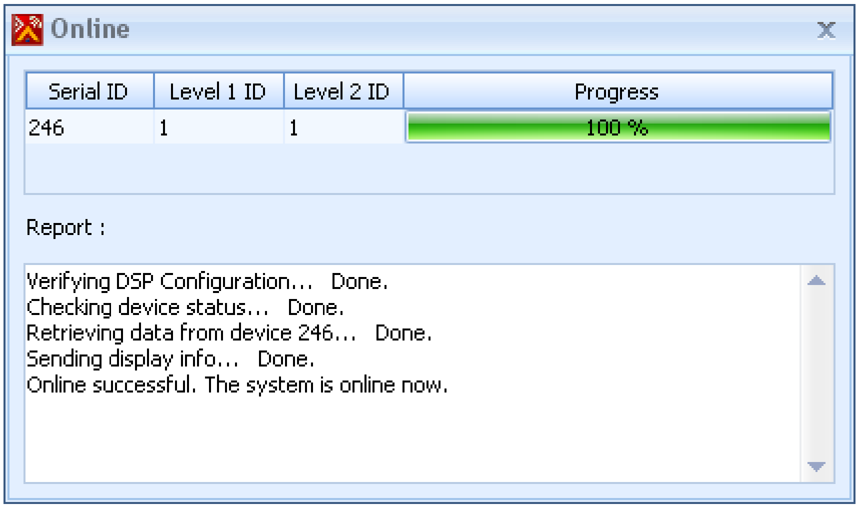


7. After this setting was finished, you can see the information on devices window show the two IDA8C are redundancy for each other. The field "Redundancy Addr" now are filled with the redundant IDA8C's address.



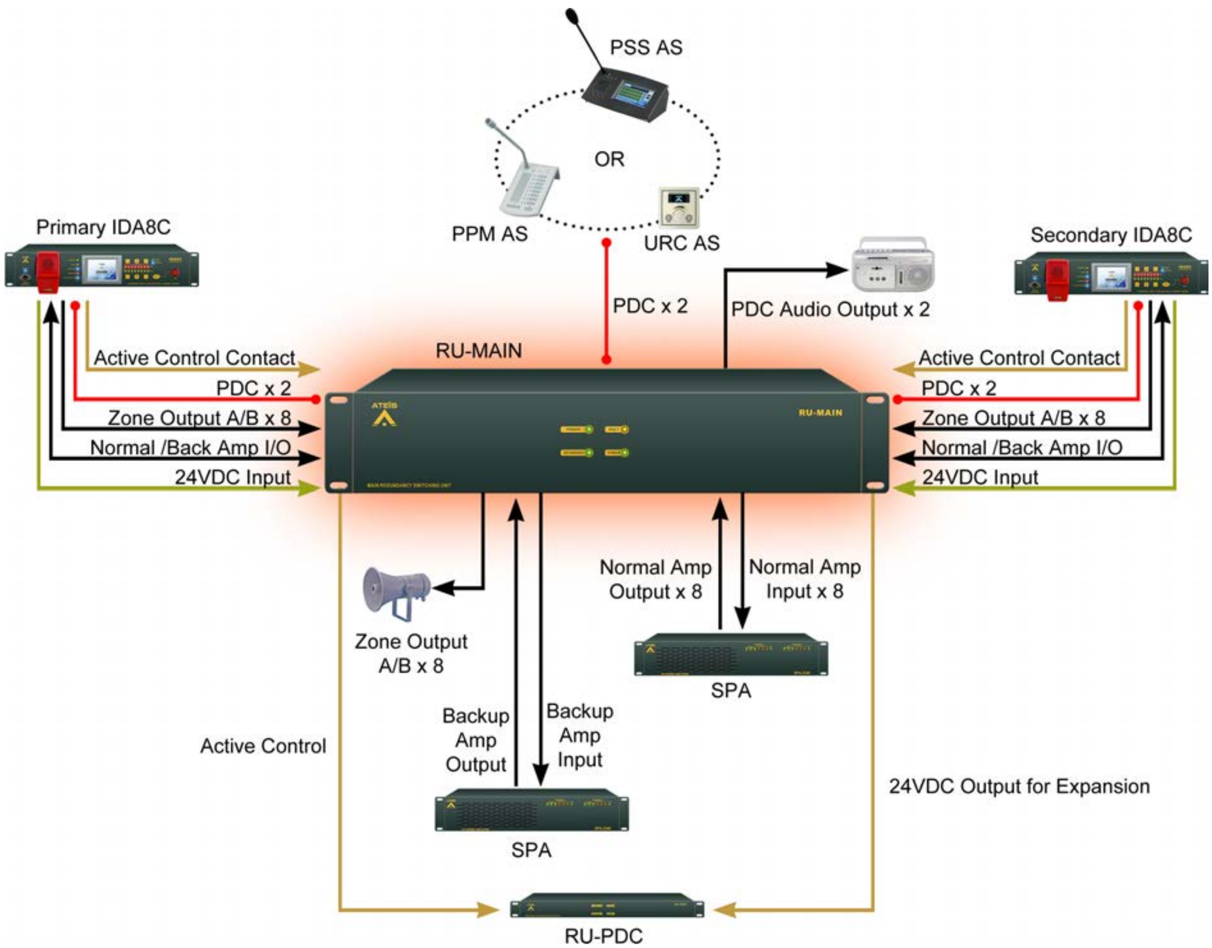
8. Store one of redundancy systems, the other will store the same configuration automatically.

9. There is a window shows the information of current system if a error detected and system switching for redundancy



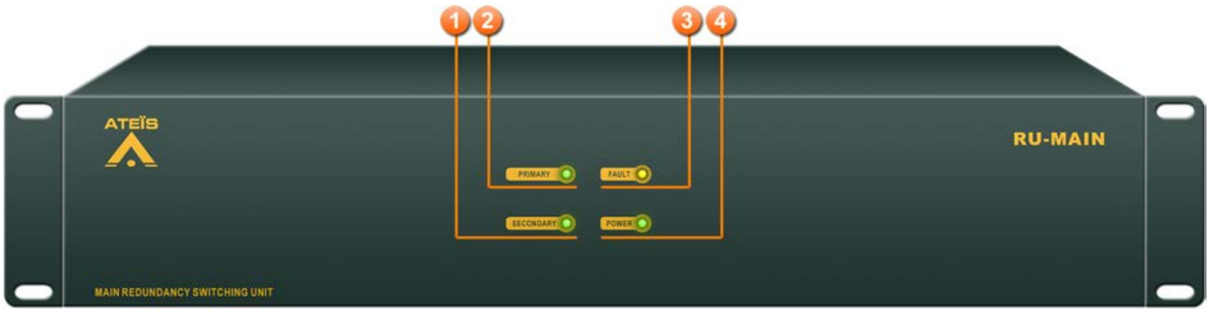
4.2.1.3 RU-Main

4.2.1.3.1 Overview



RU-MAIN is a device for audio processor redundancy, the peripherals are connected with RU-MAIN not directly to audio processor. RU-MAIN is in charge of switching primary and secondary audio processor to active one of them. If primary audio processor is active, all signal of peripherals are redirect to primary audio processor by RU-MAIN.RU-MAIN monitor the status of audio processor. if primary audio processor is crash, RU-MAIN will detected and switch to secondary audio processor.

4.2.1.3.2 Front Panel



1. Secondary Active Indicator

   This LED light up if primary audio processor is active.

2. Primary Active Indicator

   This LED light up if secondary audio processor is active.

3. Fault Indicator

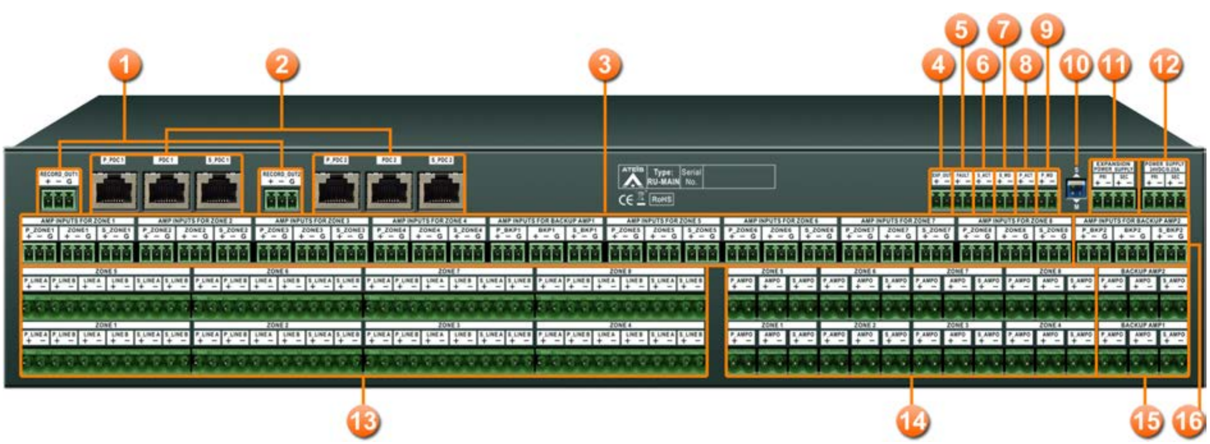
   This LED light up if the unit is set to be slave, and a short or open is detected on S. WD(Secondary Watching Dog) port.

   ⚠ If this indicator light up, it means there is an error of the unit, user needs to check wiring of S. WD connector or replace by a good unit.

4. Power Indicator

   This LED light up if this unit is power on.

4.2.1.3.3 Rear Panel



1. PDC Audio Output for Record

   The audio output of peripheral plugged on PDC connector.

2. PDC Connectors

   There are 2 sets of PDC port in RU-Main, each set consist of three connectors, allow system to do one PDC peripheral redundancy, the "N" is the number of set:

   - P_PDC N

     Connect to one of primary audio processor's PDC connectors.

   - PDC N

     Connect to peripheral which communicates to audio processor using PDC interface, for example PSS AS, PPM AS.

   - S_PDC N

     Connect to one of secondary audio processor's PDC connectors.

3. Zone Amplifier Inputs

   There are 8 sets of Zone Amplifier Input in RU-MAIN, each set consist of three connectors, allow system to do one channel of zone amplifier redundancy, the "N" is the number of set:

   - P_ZONE N

     Connect to N^{th} channel of primary audio processor's zone amplifier input connector.

   - ZONE N

     Connect to N^{th} channel of normal amplifier's audio input connector.

   - S_ZONE N

     Connect to N^{th} channel of secondary audio processor's zone amplifier input connector.

4. EXP_OUT(Active Control Expansion Output)

   This port output logic signal to next RU devices to synchronize redundancy state which tells primary or secondary is active.

5. Fault

   This port is a logic contact, which is normally closed, opened if the unit is set to be slave(refer to Master/Slave Switch of later item), and a short or open is detected on S_WD(Secondary Watching Dog) port.

6. S_ACT(Secondary Active Output)

   Output a logic signal to indicate secondary audio processor is active.

7. S_WD(Secondary Watching Dog)

   RU-Main/PDC/CTL determine active system will be primary or secondary IDA8 network by monitor input logic contacts P_WD and S_WD, The basic philosophy is that master RU monitor the signal coming from primary and secondary audio processor to decide which one can be active, and also pass the decision to next RU through EXP_OUT port. When RU is set to be master, it monitor both P_WD and S_WD for choose which IDA8 network will be active, but if RU is set to slave, it only

monitor S_WD to determine which one can be active.

8. P_ACT(Primary Active Output)

   Output a logic signal to indicate primary audio processor is active.

9. P_WD(Primary Watching Dog)

   RU devices monitor this logic signal for determining active audio processor, please refer to item "(S_WD)Secondary Watching Dog".

10. Master/Slave Switch

   A dip switch to set RU-MAIN be master or slave. The difference between master and slave please refer to item "(S_WD)Secondary Watching Dog".

11. Expansion Power Supply

   Output 24V DC power to next RU-MAIN/PDC/CTL devices.

   There are two connectors of this output:

   - PRI: Connect to Power Supply PRI connector next RU-MAIN/PDC/CTL.
   - SEC: Connect to Power Supply SEC connector next RU-MAIN/PDC/CTL.

12. Power Supply

   24V DC power input, which is connect to 24V DC Output of IDA8C/S.

   There are two connectors of this input:

   - PRI: 24V DC power input from primary audio processor.
   - SEC: 24V DC power input from secondary audio processor.

13. Speaker Zone Output

   There are 8 zones for speaker connection. each zone consists of following connectors(from left to right):

   - P_LINEA: Connect to speaker zone output line A of primary audio processor.
   - P_LINEB: Connect to speaker zone output line B of primary audio processor.
   - LINE A: Connect to line A speaker.
   - LINE B: Connect ot line B speaker.
   - S_LINE A: Connect to speaker zone output line A of secondary audio processor.
   - S_LINE B: Connect to speaker zone output line B of secondary audio processor.

14. Zone Amplifier Output

   There are 8 zones for amplifier output connection, it receive the gained audio signal coming from normal amplifier, each amplifier output consist of three connectors:

   - P_AMPO: Connect to normal amplifier output connector of primary audio processor.
   - AMPO: Connect to output of normal amplifiers.
   - S_AMPO: Connect to normal amplifier output connector of secondary audio processor.

15.Backup Amplifier Output

There are two backup amplifier outputs on RU-MAIN, each output consist of three connectors:

- P_AMPO: Connect to backup amplifier output of primary audio processor.
- AMPO: Connect to backup amplifier output.
- S_AMPO: Connect to backup amplifier output of secondary audio processor.

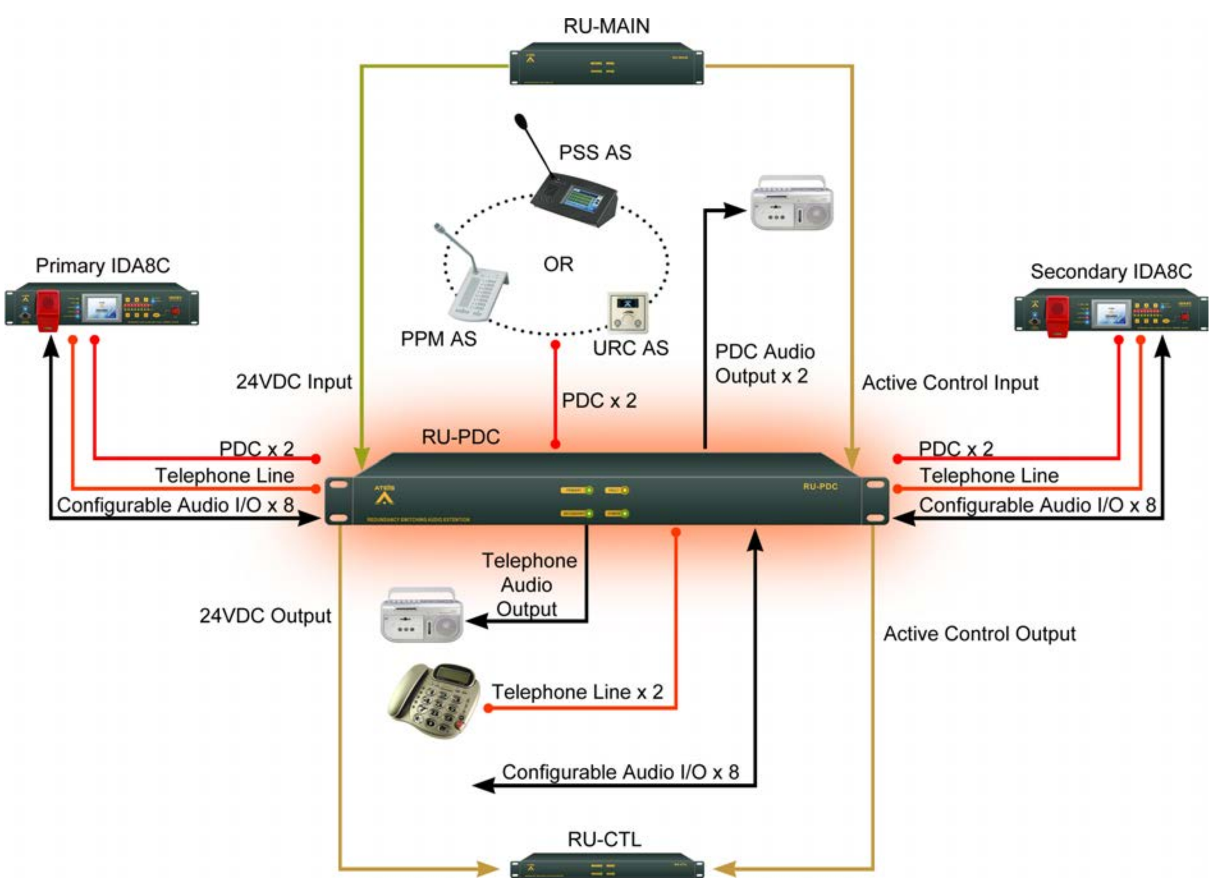
16.Backup Amplifier Input

There are two backupt amplifier inputs on RU-MAIN, each input consist of three connectors, where N is the number of channel:

- P_BKP N: Connect to backup amplifier input of primary audio processor.
- BKP N: Connect to input of backup amplifier.
- S_BKP N: Connect to backup amplifier input of secondary audio processor.

4.2.1.4 RU-PDC

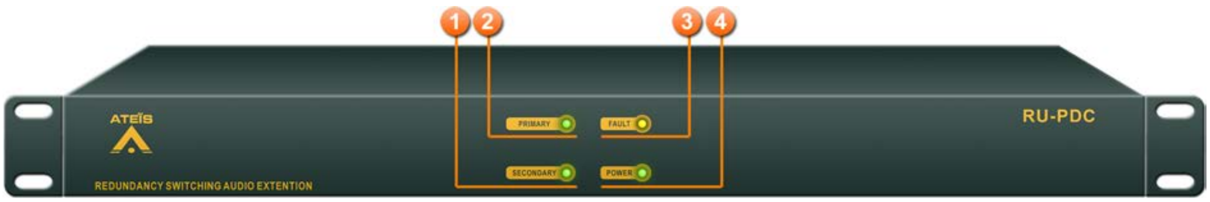
4.2.1.4.1 Overview



RU-PDC is a device for audio processor redundancy, the peripherals are connected with RU-PDC not directly to audio processor. RU-PDC is in charge of switching primary and secondary audio processor to

active one of them. If primary audio processor is active, all signal of peripherals are redirect to primary audio processor by RU-PDC. RU-PDC monitor the status of the other RUs for switching active audio processor.

4.2.1.4.2 Front Panel



1. Secondary Active Indicator

   This LED light up if active system is primary audio processor.

2. Primary Active Indicator

   This LED light up if active system is secondary audio processor.

3. Fault Indicator

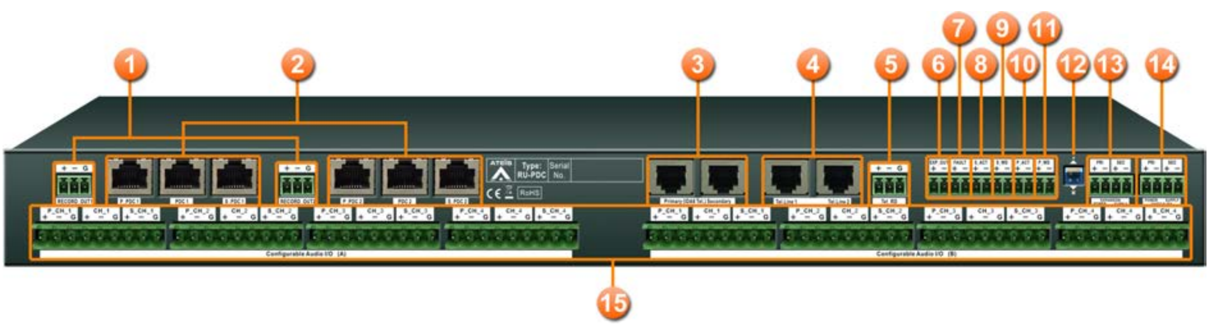
   This LED light up if the unit is set to be slave, and a short or open is detected on S. WD(Secondary Watching Dog) port.

   If this indicator light up, it means there is an error of the unit, user needs to check wiring of S. WD connector or replace by a good unit.

4. Power Indicator

   This LED light up if this unit is power on.

4.2.1.4.3 Rear Panel



1. PDC Audio Output for Record

   The audio output of peripheral plugged on PDC connector.

2. PDC(Peripherals Device Controller) Connectors

   There are 2 sets of PDC port in IDA8RU-PDC, each set consist of three connectors, allow system to do one PDC peripheral redundancy, the "N" is the number of set:

   - P_PDC N

     Connect to one of primary audio processor's PDC connectors.

- PDC N

  Connect to peripheral which communicates to audio processor using PDC interface, for example PSS AS, PPM AS.

- S_PDC N

  Connect to one of secondary audio processor's PDC connectors.

3. IDA8 Telephone Connector

   There are two connectors for IDA8 telephone line:

   - Primary Audio Processor Telephone: Connect to telephone line connector of primary audio processor.
   - Secondary Audio Processor Telephone: Connect to telephone line connector of secondary audio processor.

4. Telephone Line Connector

   There are two connectors for telephone line of telephone company:

   - Tel. Line1: Line1 for telephone company connection.
   - Tel. Line2: Line2 for telephone company connection.

5. Telephone Line Audio Output for Record

   Audio output of active telephone line.

6. Active Control Expansion Output

   This port output logic signal to next RU devices to synchronize redundancy state which tells primary or secondary is active.

7. Fault

   This port is a logic contact, which is normally closed, opened if the unit is set to be slave(refer to Master/Slave Switch of later item), and a short or open is detected on S_WD(Secondary Watching Dog) port.

8. S_ACT(Secondary Active Output)

   Output a logic signal to indicate secondary audio processor is active.

9. S_WD(Secondary Watching Dog)

   IDA8RU-Main/PDC/CTL determine active system will be primary or secondary audio processor by monitor input logic contacts P_WD and S_WD, The basic philosophy is that master RU monitor the signal coming from primary and secondary audio processor to decide which one can be active, and also pass the decision to next RU through EXP_OUT port. When RU is set to be master, it monitor both P_WD and S_WD for choose which audio processor will be active, but if RU is set to slave, it only monitor S_WD to determine which one can be active.

10. P_ACT(Primary Active Output)

    Output a logic signal to indicate primary audio processor is active.

11. P_WD(Primary Watching Dog)

    RU devices monitor this logic signal for determining active audio processor, please refer to item

"(S_WD)Secondary Watching Dog".

12.Master/Slave Switch

A dip switch to set RU-PDC be master or slave. The difference between master and slave please refer to item "(S_WD)Secondary Watching Dog".

13.Expansion Power Supply

Output 24V DC power to next RU-MAIN/PDC/CTL devices.

There are two connectors of this output:

- PRI: Connect to Power Supply PRI connector next RU-MAIN/PDC/CTL.
- SEC: Connect to Power Supply SEC connector next RU-MAIN/PDC/CTL.

14.Power Supply

24V DC power input, which is connect to 24V DC Output of audio processor.

There are two connectors of this input:

- PRI: 24V DC power input from primary audio processor.
- SEC: 24V DC power input from secondary audio processor.

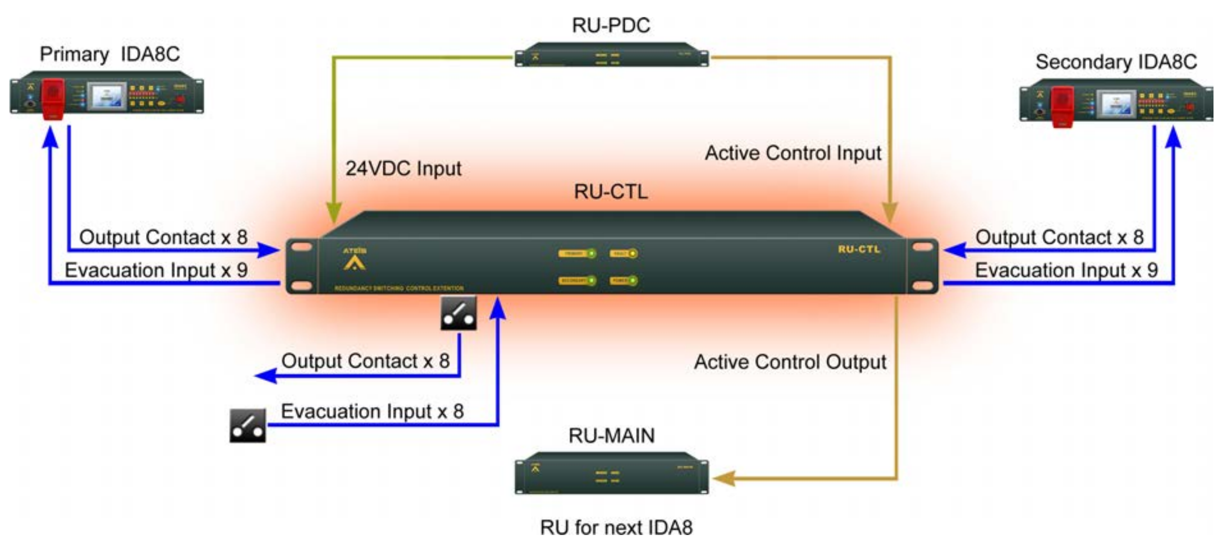
15.Configurable Audio I/O

There are two ports(A/B) for configurable audio I/O, each port contains four channels, each channel consist of three connectors, where postfix "N" is the number of channel:

- P_CH_N: Connect to configurable audio I/O connector of primary audio processor.
- CH_N: Connect to audio In/Out equipment.
- S_CH_N: Connect to configurable audio I/O connector of secondary audio processor.

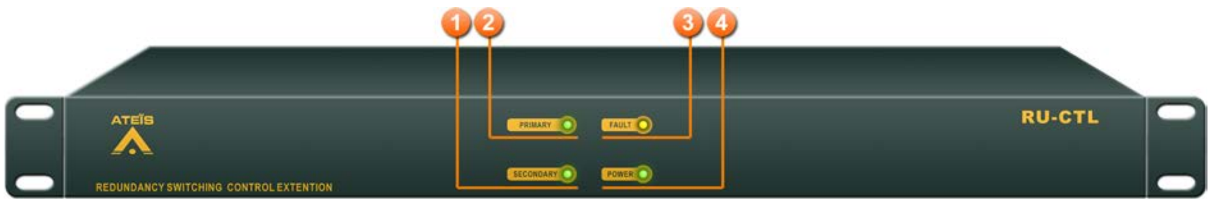
4.2.1.5 RU-CTL

4.2.1.5.1 Overview



RU-CTL is a device for audio processor redundancy, the peripherals are connected with RU-CTL not directly to audio processor. RU-CTL is in charge of switching primary and secondary audio processor to active one of them. If primary audio processor is active, all signal of peripherals are redirect to primary audio processor by RU-CTL. RU-CTL monitor the status of the other RUs for switching active audio processor.

4.2.1.5.2 Front Panel



1. Secondary Active Indicator

   This LED light up if active system is primary audio processor.

2. Primary Active Indicator

   This LED light up if active system is secondary audio processor.

3. Fault Indicator

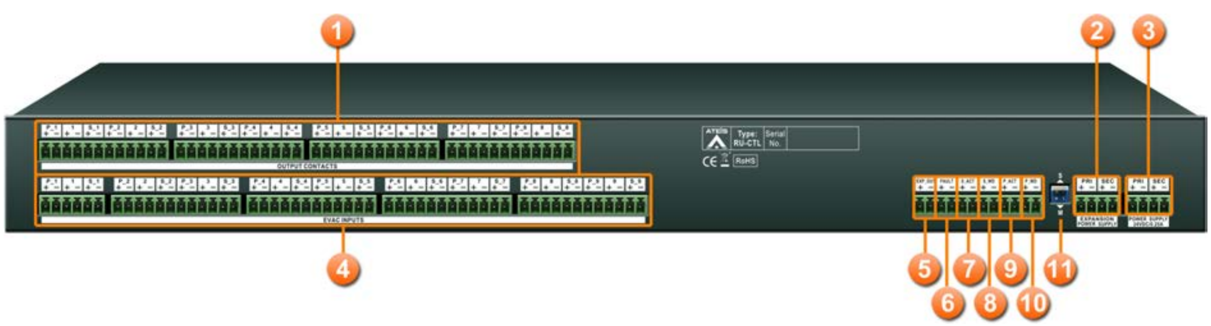
   This LED light up if the unit is set to be slave, and a short or open is detected on S. WD(Secondary Watching Dog) port.

   If this indicator light up, it means there is an error of the unit, user needs to check wiring of S. WD connector or replace by a good unit.

4. Power Indicator

   This LED light up if this unit is power on.

4.2.1.5.3 Rear Panel



1. Output Contacts

   There are 8 channels for output contacts, each channel consist of three connectors, where "N" is the number of channel:

   - P_N: Connect to Nth output contact of primary audio processor.
   - N: Nth channel of output contact.
   - S_N: Connect to Nth output contact of secondary audio processor.

2. Expansion Power Supply

   Output 24V DC power to next RU-MAIN/PDC/CTL devices.

   There are two connectors of this output:

   - PRI: Connect to Power Supply PRI connector next RU-MAIN/PDC/CTL.
   - SEC: Connect to Power Supply SEC connector next RU-MAIN/PDC/CTL.

3. Power Supply

   24V DC power input, which is connect to 24V DC Output of audio processor.

   There are two connectors of this input:

   - PRI: 24V DC power input from primary audio processor.
   - SEC: 24V DC power input from secondary audio processor.

4. Evacuation Inputs

   There are 9 channels for evacuation input, each channel consist of three connectors, where "N" is the number of channel:

   - P_N: Connect to N^{th} evacuation input of primary audio processor.
   - N: N^{th} channel of evacuation input.
   - S_N: Connect to N^{th} evacuation input of secondary audio processor.

5. Active Control Expansion Output

   This port output logic signal to next RU devices to synchronize redundancy state which tells primary or secondary is active.

6. Fault

   This port is a logic contact, which is normally closed, opened if the unit is set to be slave(refer to Master/Slave Switch of later item), and a short or open is detected on S_WD(Secondary Watching Dog) port.

7. S_ACT(Secondary Active Output)

   Output a logic signal to indicate if secondary IDA8 network is active.

8. S_WD(Secondary Watching Dog)

   IDA8RU-Main/PDC/CTL determine active system will be primary or secondary IDA8 network by monitor input logic contacts P_WD and S_WD, The basic philosophy is that master RU monitor the signal coming from primary and secondary audio processor to decide which one can be active, and also pass the decision to next RU through EXP_OUT port. When RU is set to be master, it monitor both P_WD and S_WD for choose which audio processor will be active, but if RU is set to slave, it only monitor S_WD to determine which one can be active.

9. P_ACT(Primary Active Output)

   Output a logic signal to indicate if primary audio processor is active.

10. P_WD(Primary Watching Dog)

   RU devices monitor this logic signal for determining active audio processor, please refer to item

"(S_WD)Secondary Watching Dog".

11.Master/Slave Switch

A dip switch to set RU-PDC be master or slave. The difference between master and slave please refer to item "(S_WD)Secondary Watching Dog".

4.2.2 DNM

4.2.2.1 Overview

The DNM assures the omnidirectional sound recording and preamplification of the surrounding background noise.

The 0 dB modulation is sent through the Audio Processor in order to provide the automatic gain control feature of the DNM component and allows the level adjustment where the DNM is implemented.

Any situation in which the volume of loudspeakers need to be adjusted automatically depending on the background noise. (ex: station, Airport, MRT, etc.)

4.2.2.2 Installation

Install the DNM device to the position in which you want the DNM to sense background noise and adjust loudspeaker volume automatically, and connect the DNM device with Audio Processor by RS485 or Ethernet, besides, if the system needs to connect to a series of DNM on same RS485 port, then the user should plug in the DNM devices one by one before connecting a series DNM to RS485 port. How to connect is described in the "Configuration" section.

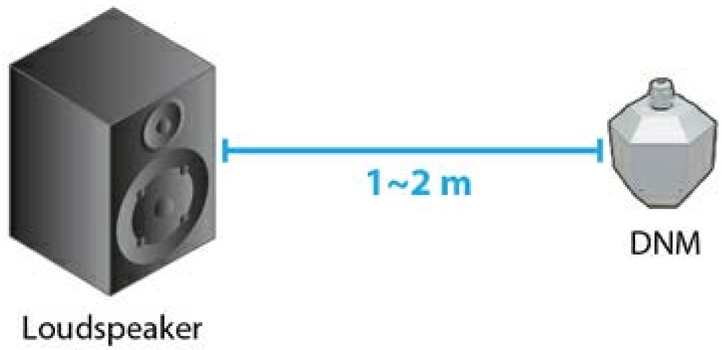
One DNM can only support one Zone(like IDA8 Zone Channel).

Generally, place the DNM in front of a loudspeaker about 1~2 meters. The distance between DNM and Loudspeaker also depends on the power of the amplifier and loudspeaker.

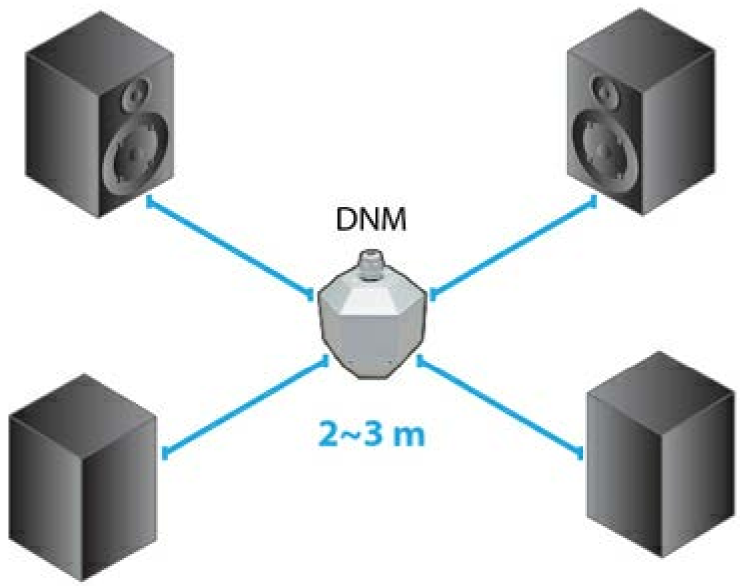
If it is a high-power loudspeaker (sound is louder), then the distance should be increased, and vice versa.

❖ The examples of installation as follows:

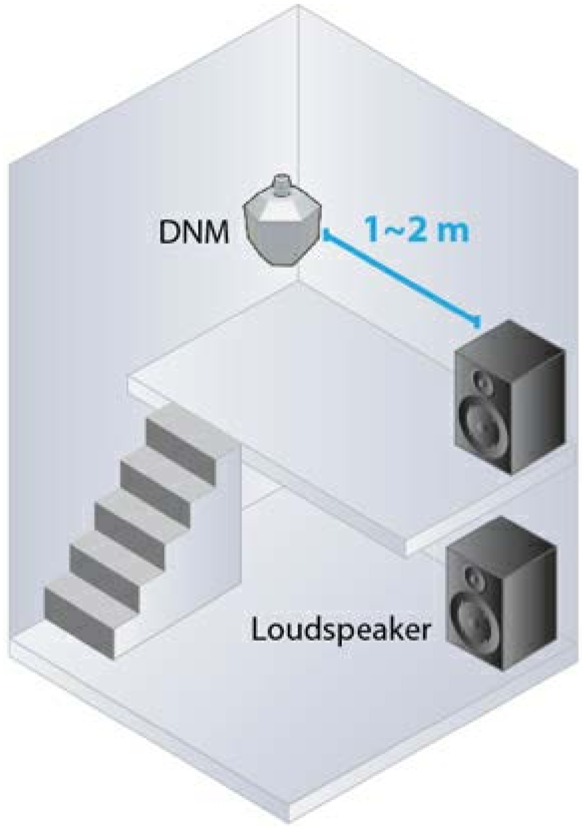
1. One zone connects only one loudspeaker, generally, then install the DNM in front of the loudspeaker about 1~2 meters.



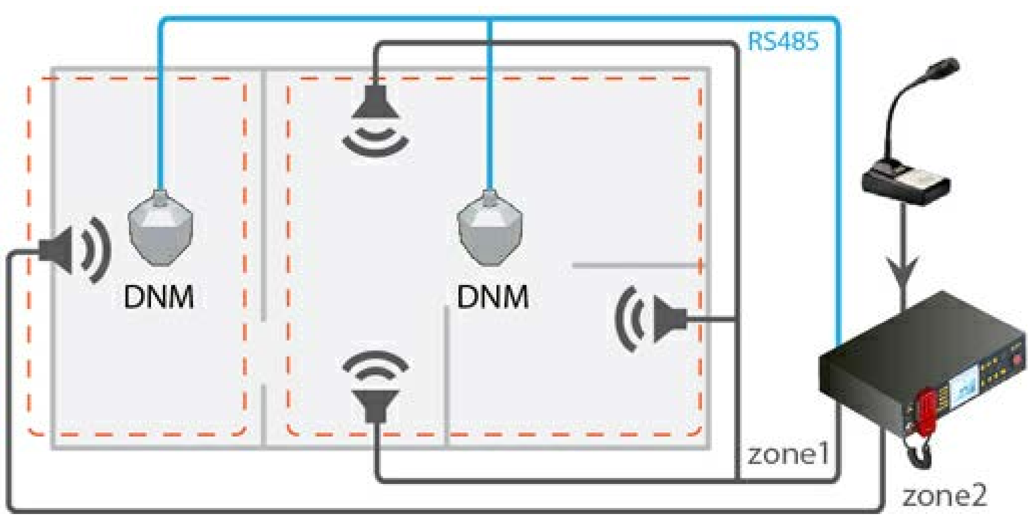
2. One zone connects many loudspeakers, and installed the loudspeakers by circularity and the distance is not far between the loudspeakers, then the DNM can be set in the center of loudspeakers (ex: A small conference room, etc.).

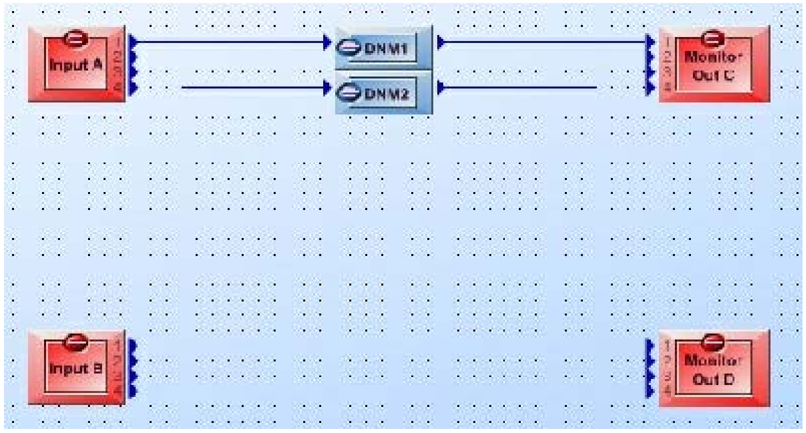


3. One zone connects many loudspeakers, and the installation of loudspeakers are irregular but the distance of all loudspeakers are not far, then install the DNM about 1~2 meters in the front of any one loudspeaker.



4. In a large area, the broadcast source is same in the area, and the distance of each loudspeaker is far, maybe the ambient noise is loud in one side, but quiet in the other side. What if the DNM is installed around loud side? The DNM will adjust the level automatically allowing people to hear the broadcast clearly, as all of the loudspeakers are connected from same source (same zone output). Maybe the volume of broadcast is too loud for the quiet side, or on the contrary, if the DNM is installed around quiet side, the volume of broadcast is too small for the loud side. To avoid this problem, the user can separate the area to several zones and one DNM for one separated zone, and all of the DNM are same input source (ex. Department store, etc.).





Note: The sound of loudspeakers from another zone can't effect each other, otherwise, the DNM will consider the sounds from another zone as a background noise.

4.2.2.3 Characteristics

❖ Case

Dimension = 100mm(Ö) x 130mm(H)

Weight = 0.13 Kg

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	250mA	-

❖ Microphone

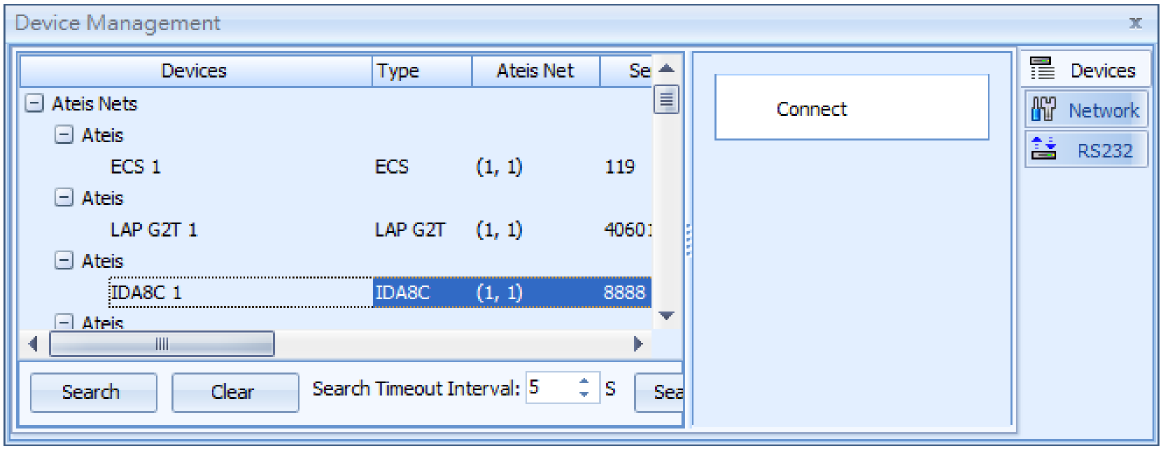
- Sensitivity adjustment rang ... (60 ~ 120dBA)±5dBA
- THD @ 1kHz < 0.2%.
- Bandwidth @ -3dB = 50Hz ~ 16kHz.

❖ Maximum Cable Length

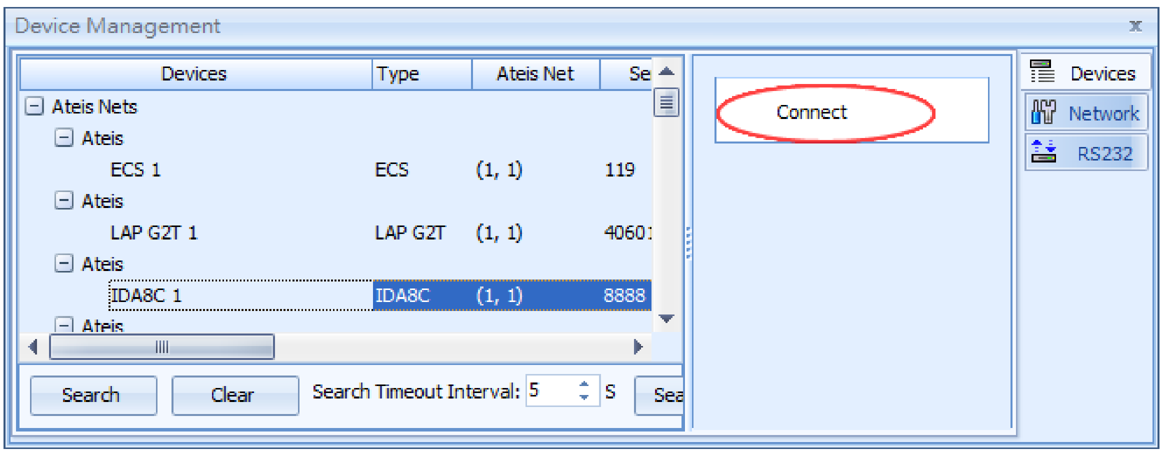
600M on category 5 cable.

4.2.2.4 Configuration

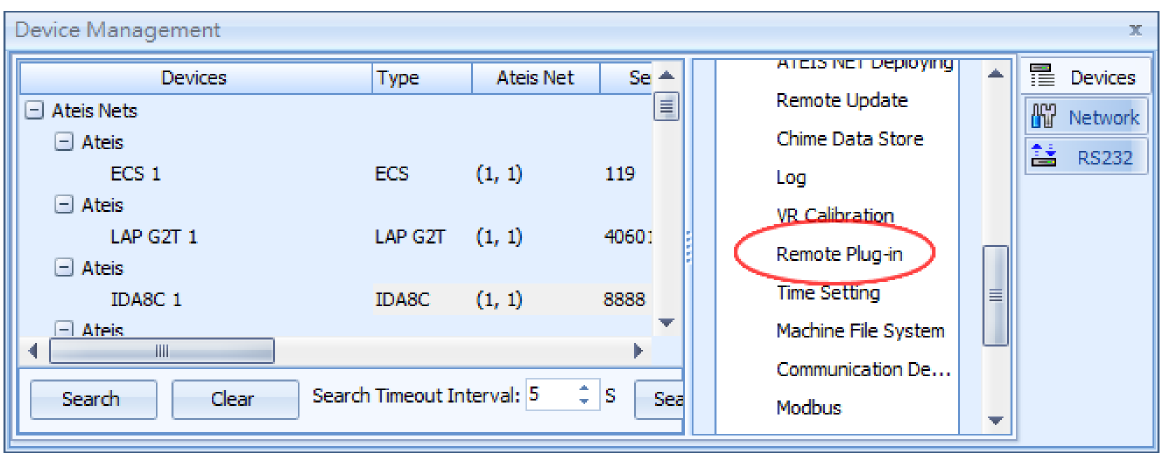
1. Plug the DNM to the RS485(PDC) port of IDA8C/S (Here's the example using IDA8, the configuration is the same for other Ateis Audio Processor).
2. Search and click on the entry of IDA8C/S in the [Device Management] window:



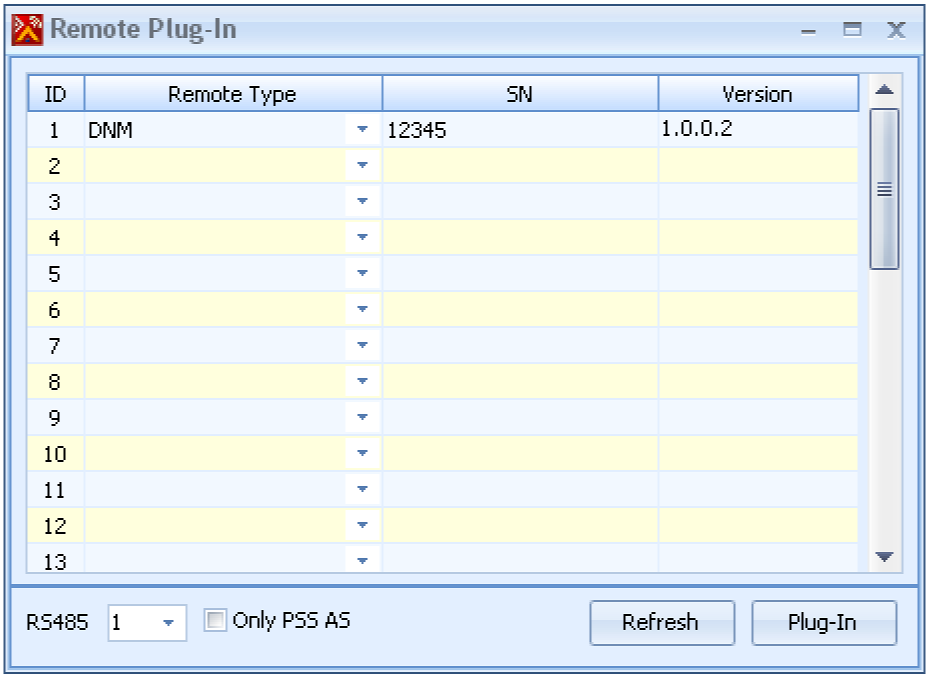
3. Connect to the IDA8C/S



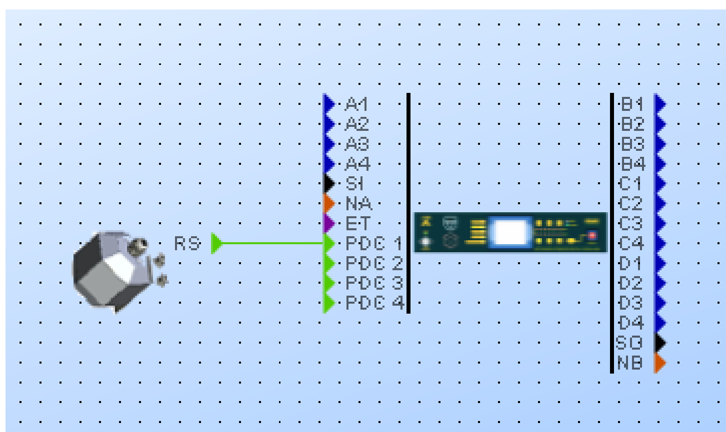
4. Click the button [Remote Plug-In]



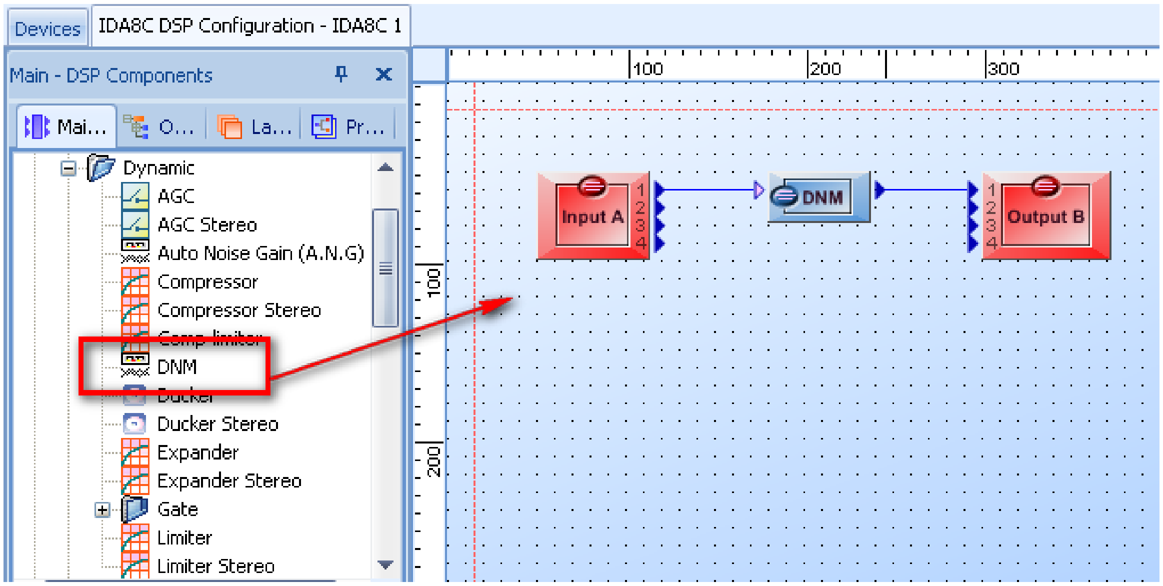
5. A window [Remote Plug-In] pop up. Select RS485 port number and Remote Type, and key in a series number (which is different with another DNM). Click the “Plug-In” button in the window. After plug in, click the “refresh” button to read back the DNM information from the RS485 port If the information is same with your plug-in, the plug-in is successful.



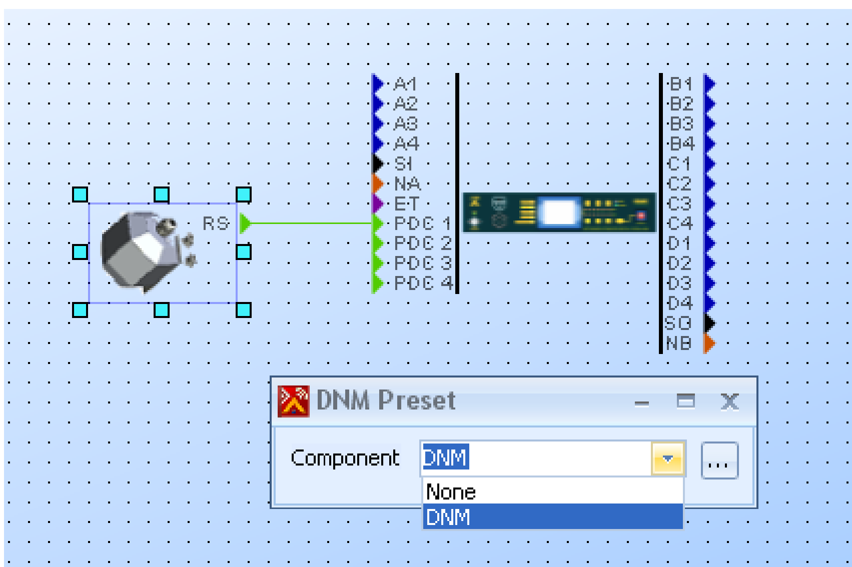
6. In device editing window, create a DNM and wires it to IDA8C/S's RS485 port:



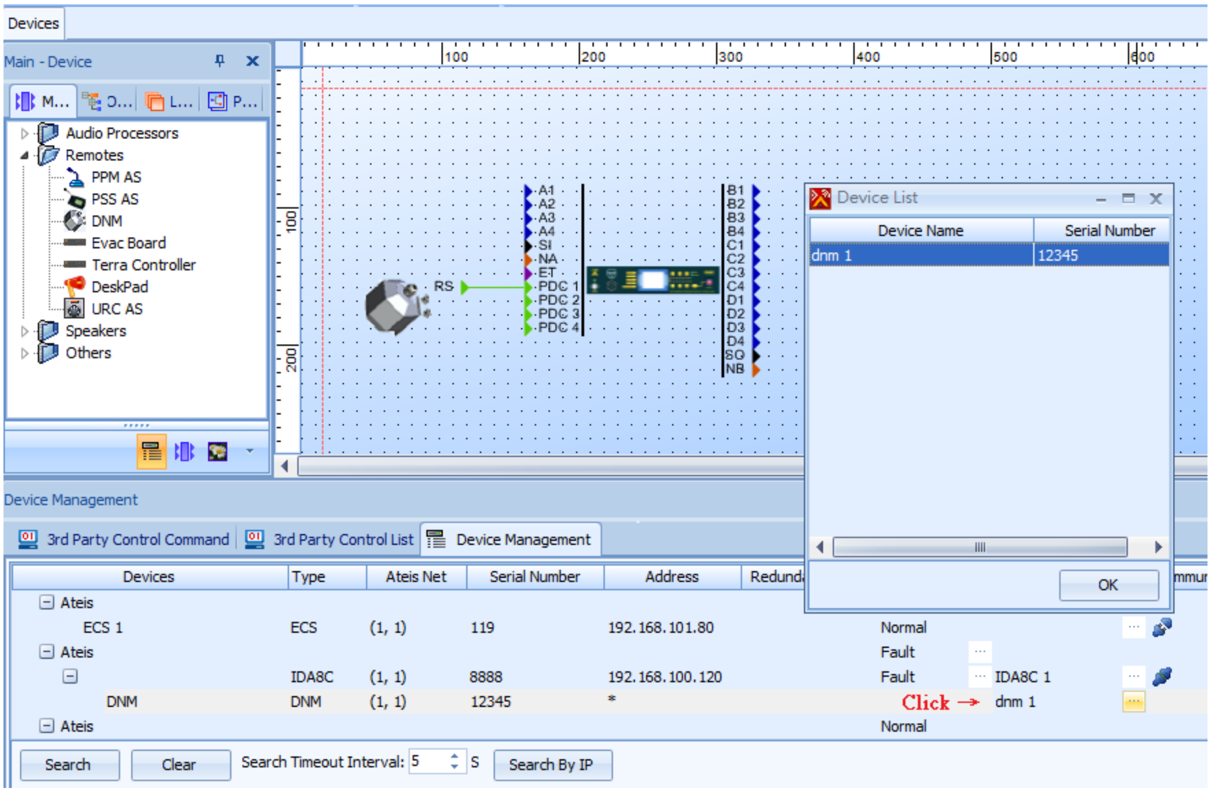
7. Double click the IDA8C/S to open DSP component editing window, then drag and drop to create "DNM" component. In this case, it simply input audio from "Input" component and output to "Output" component:



8. Go back to device editing window, click on the block of DNM device. A [DNM Preset] window pops up. There is a combo box to select the cooperate DSP component for the DNM device.

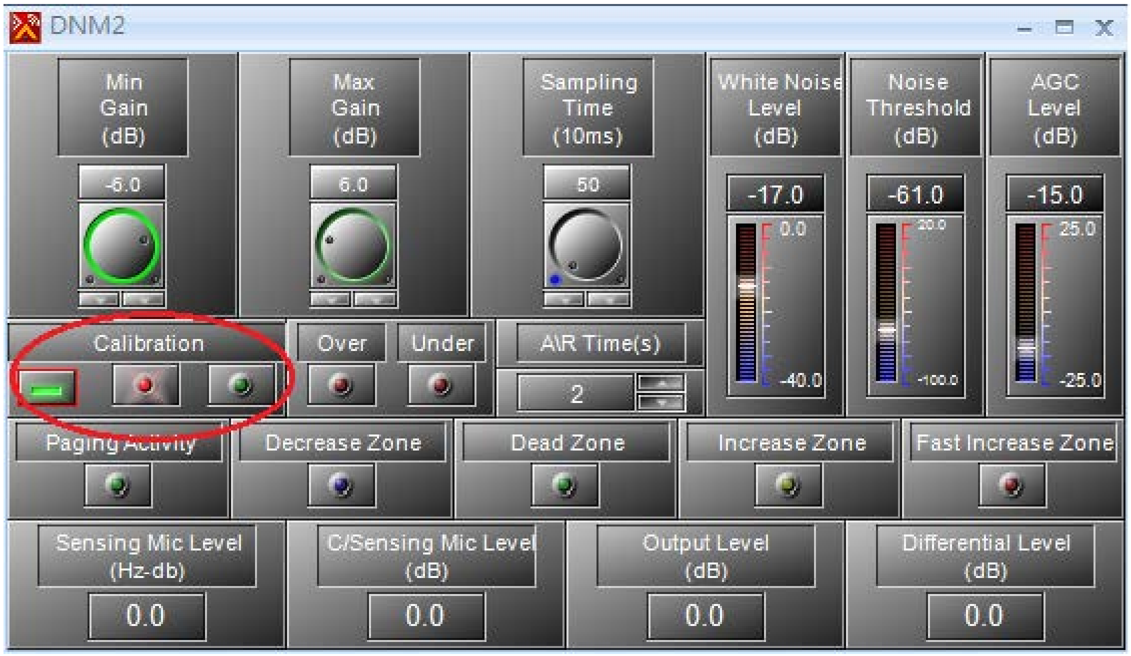


9. To assign the DNM component to device.

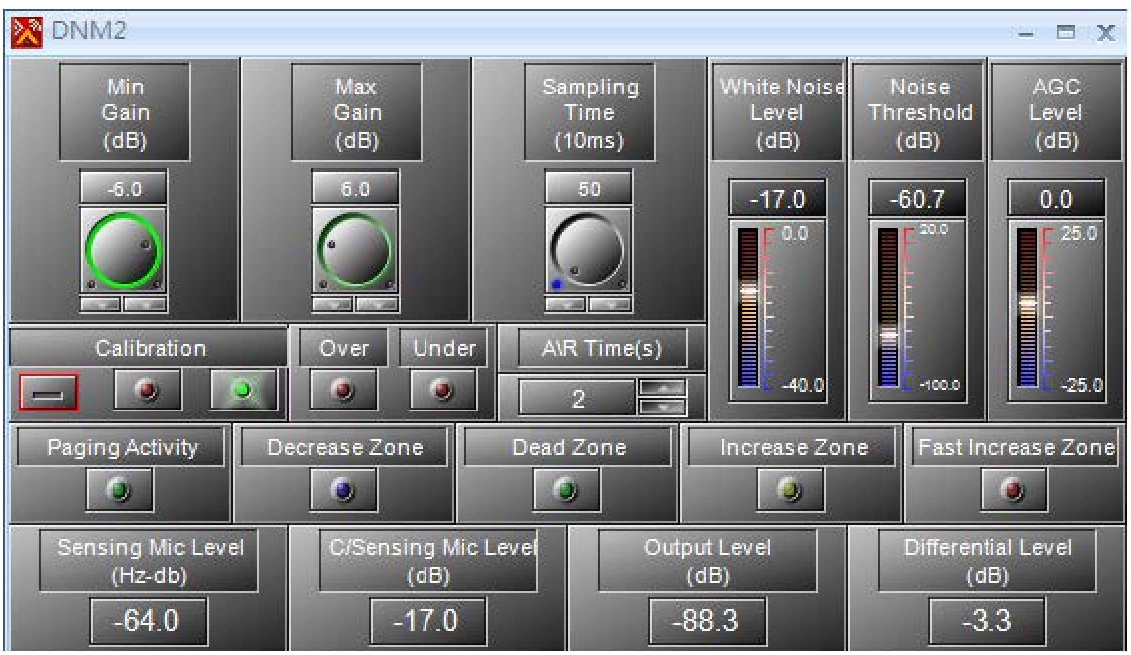


10.Store the configuration, and then online.

11.Click "DNM" component to open it's element window, press button [Calibration] to start the calibration procedure. When it is calibrating, the red LED will light up, and the White Noise Level(dB) will be auto-tuning (sending out a low frequency tone). **Note:** White Noise Level(dB) is appropriate to stop around -20dB.

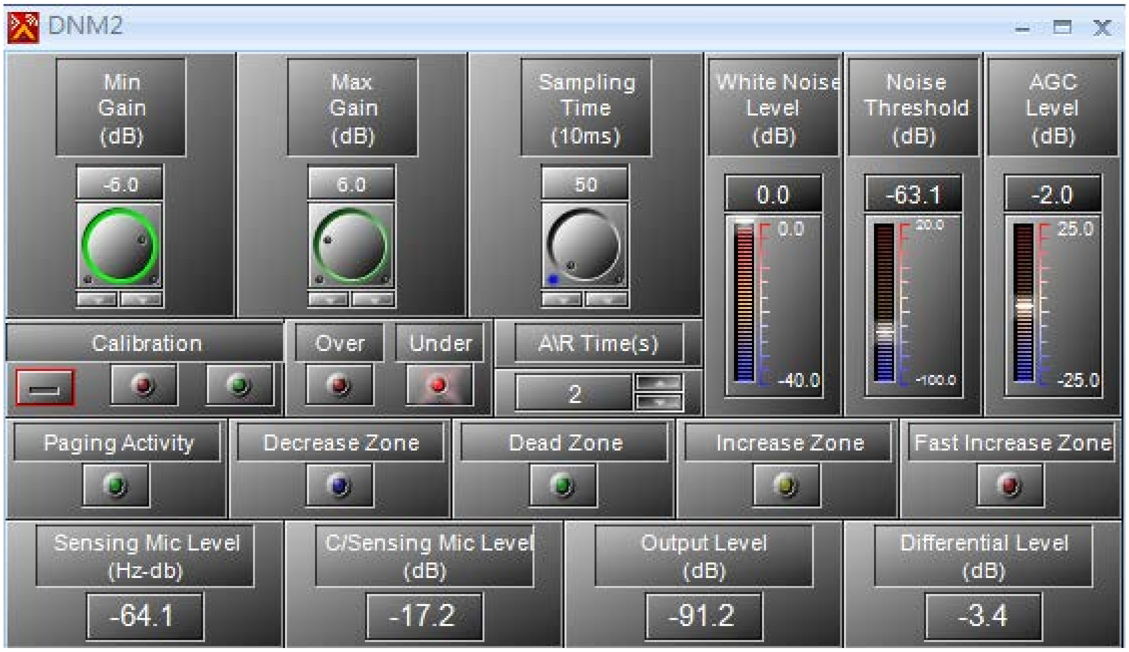


12.After the calibration is successful, the green LED will light up, and the DNM will start to work.



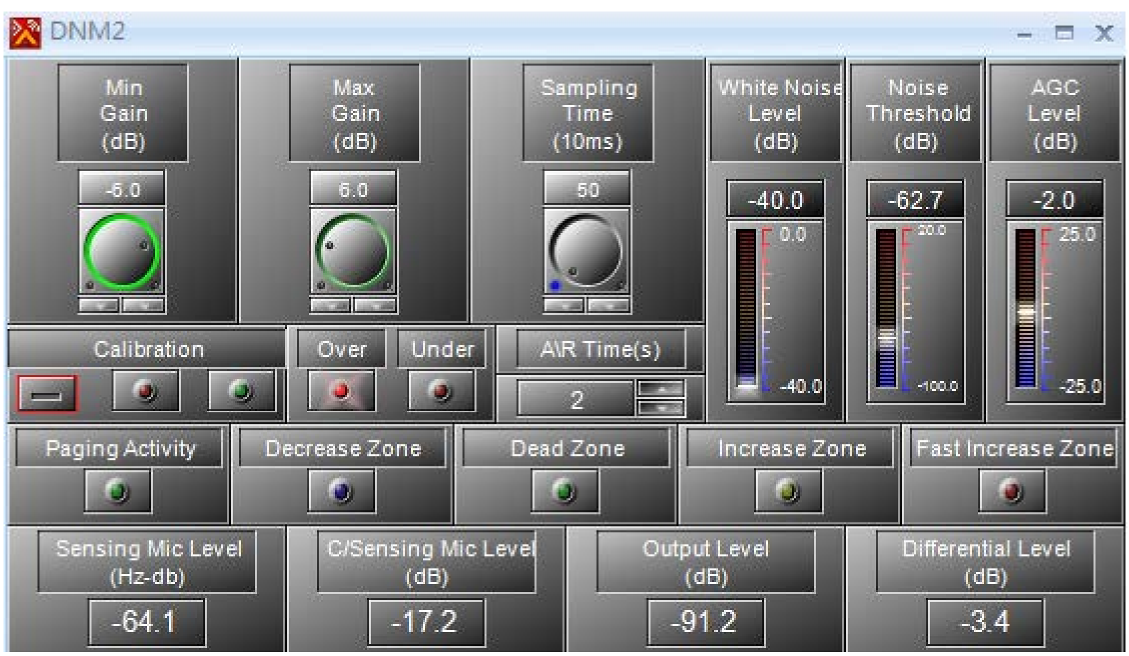
13.If the White Noise Level goes up to 0dB and the 'Under LED' is lighted up, the calibration has failed at this time, and the DNM can't receive the calibration noise coming from loudspeaker, please check:

- Is the sound of loudspeaker is too small?
- Is the distance too far between DNM and loudspeaker?
- Is the DNM damaged?



14. If the White Noise Level goes down to -40dB and the over LED is lighted up, this means the calibration has failed at this time, and the DNM received the sound too loud, please check:

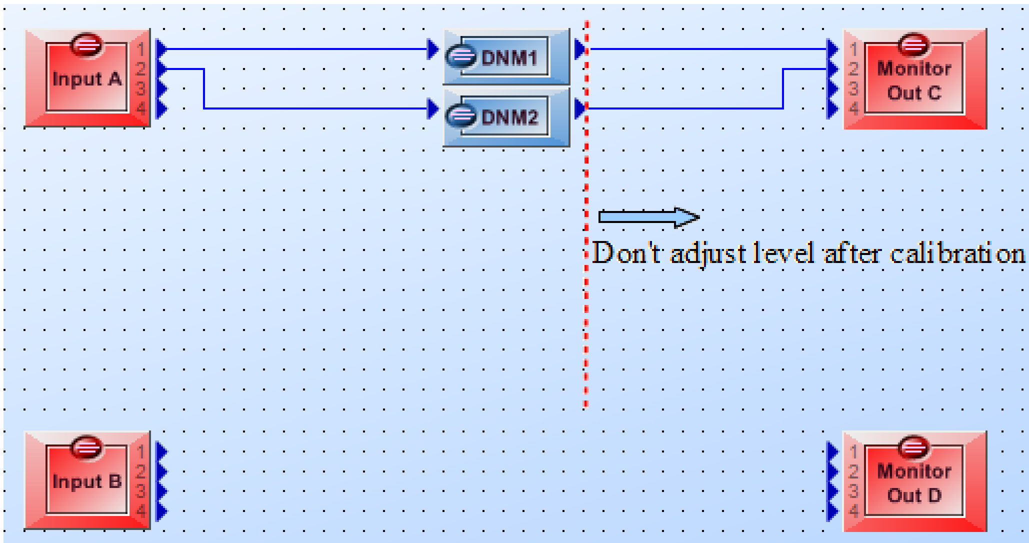
- Is The sound of loudspeaker is too loud?
- Is the distance too close between DNM and loudspeaker?
- Is the DNM damaged ?



15. After calibration finished, DNM starts to work, It increases the level of audio if there is some noise detected by DNM.

4.2.2.5 Operation Notice

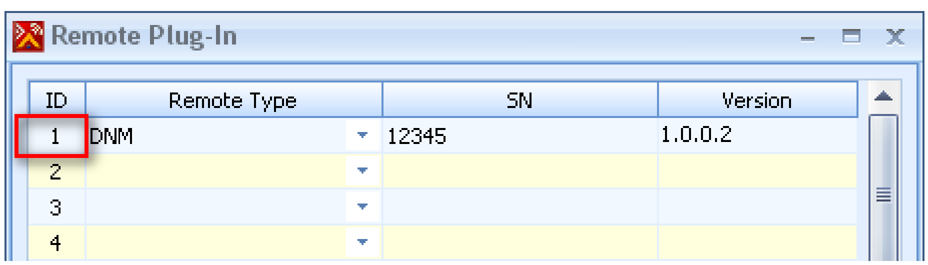
Once the calibration is successful, don't adjust the level in the rear of DNM component (ex: Monitor Out C's level or amplifier gain, etc.), but the level can be adjusted which is front of the DNM component (ex: Input A level or PC output volume, etc.).

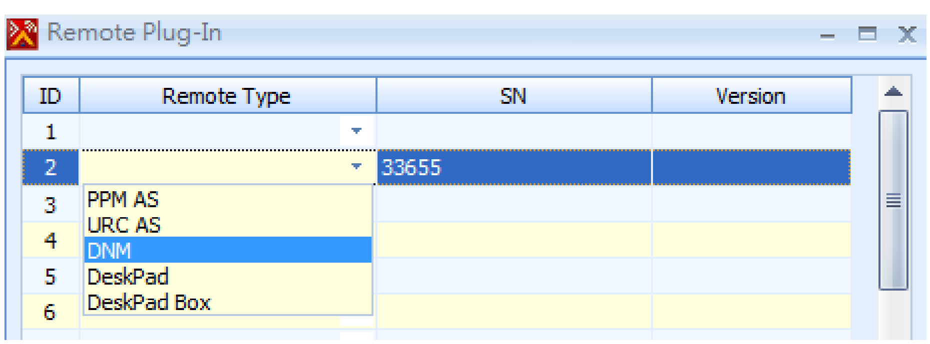


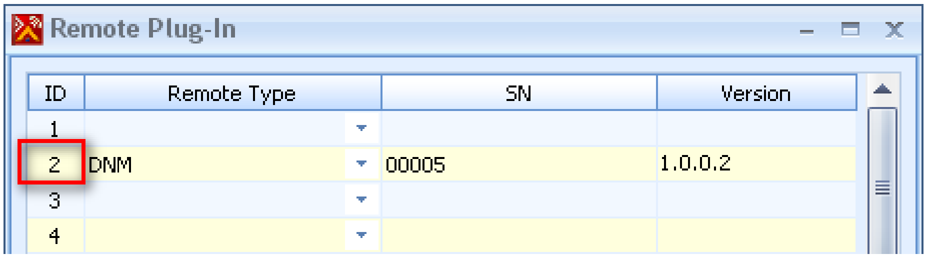
4.2.2.6 Two DNM on the Same PDC

If the system needs to use more than one DNM, and to connect a DNM series on same RS485 port, then the user should plug in the DNM one by one connecting a DNM series to RS485 port. The steps of plug-in is same as above. Note: The DNM should be connected to a different ID and Series number.

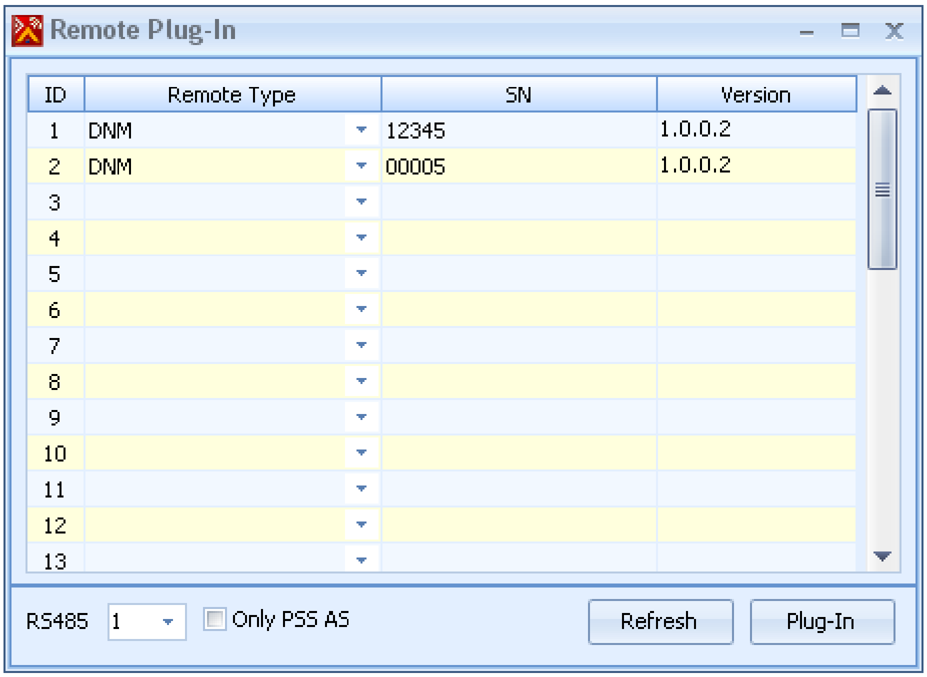
1. For each DNM, plug into the RS485 port of IDA8C/S, and assign an ID for it.



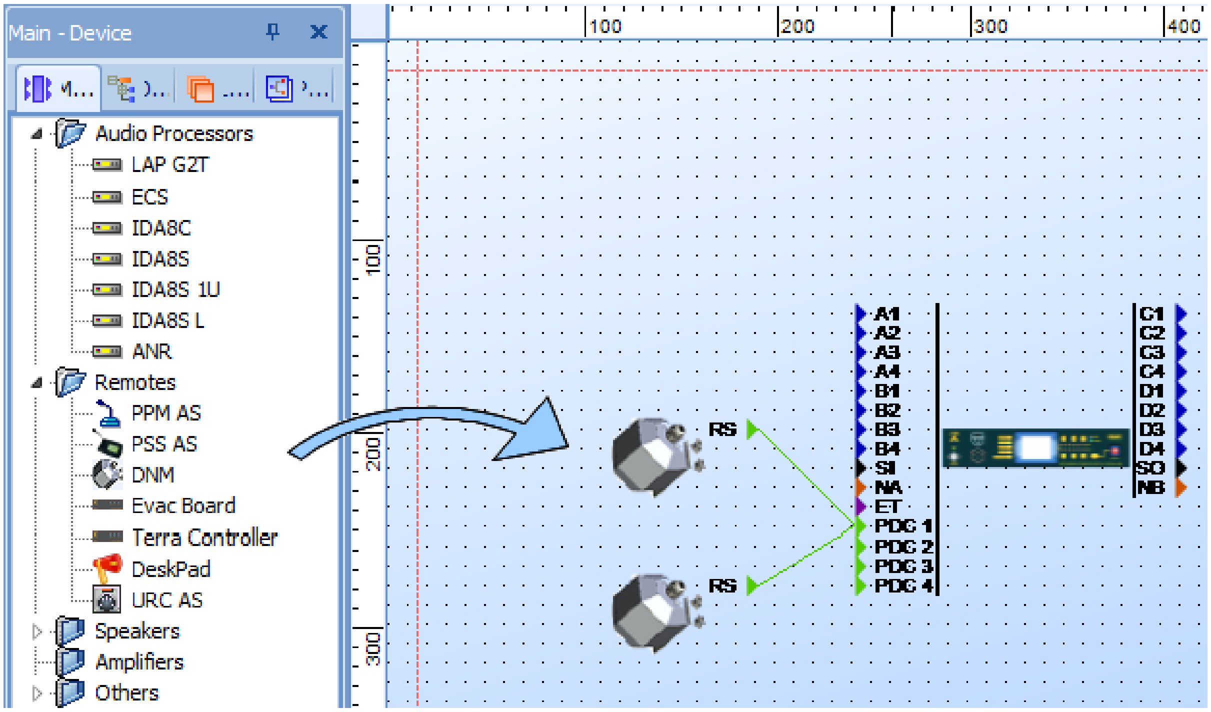




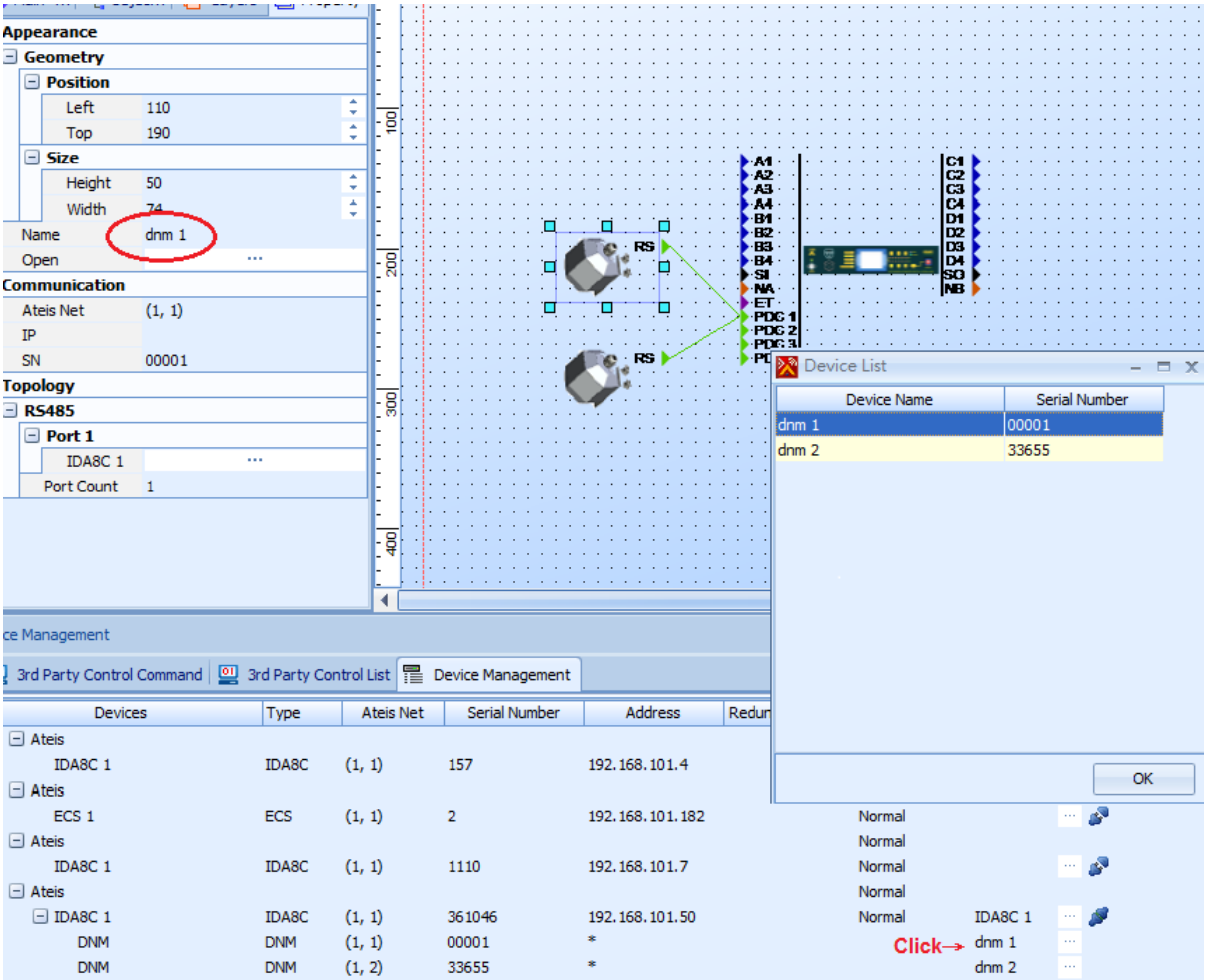
2. After connecting two DNM one by one, then connect the DNM to a series. Click the "Refresh" button to read back the information, now these two DNMs are listed on the grid.



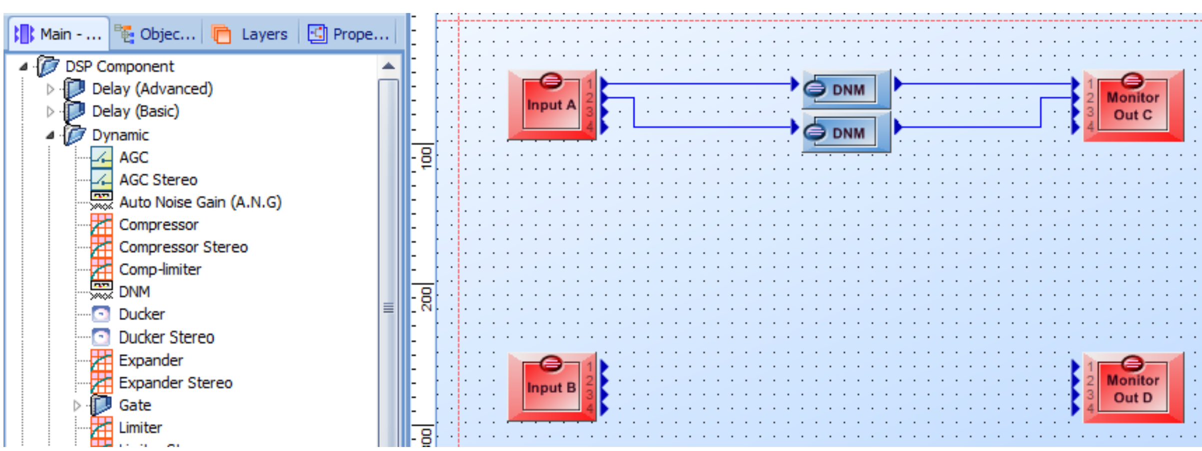
3. In the configuration file, draw the DNM component out, and connect to RS485 port (same with real wiring). If the system has two DNM, then draw two DNM device components.(the example is two DNM in PDC1).



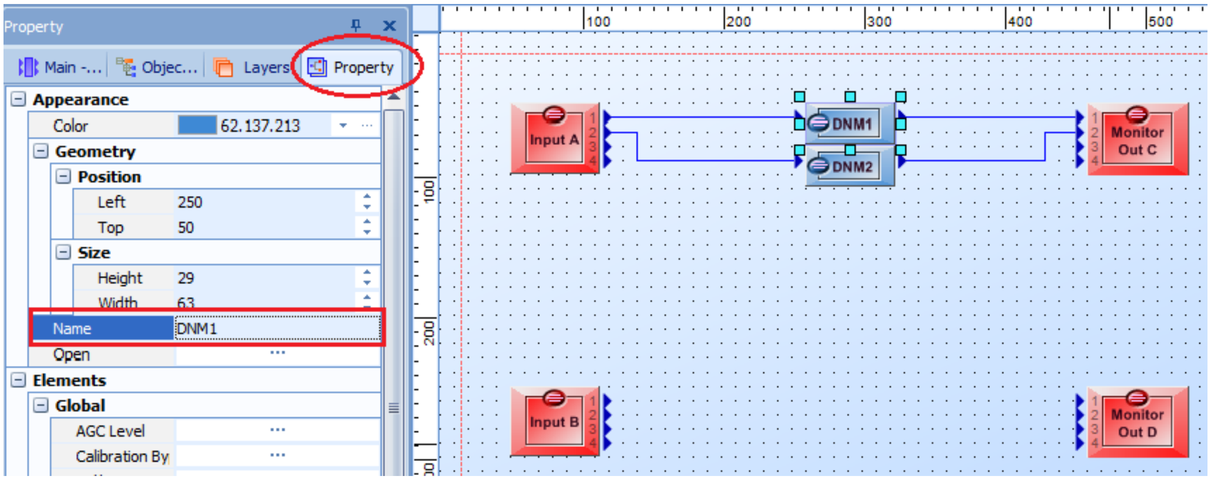
4. To assign the DNM component to device.



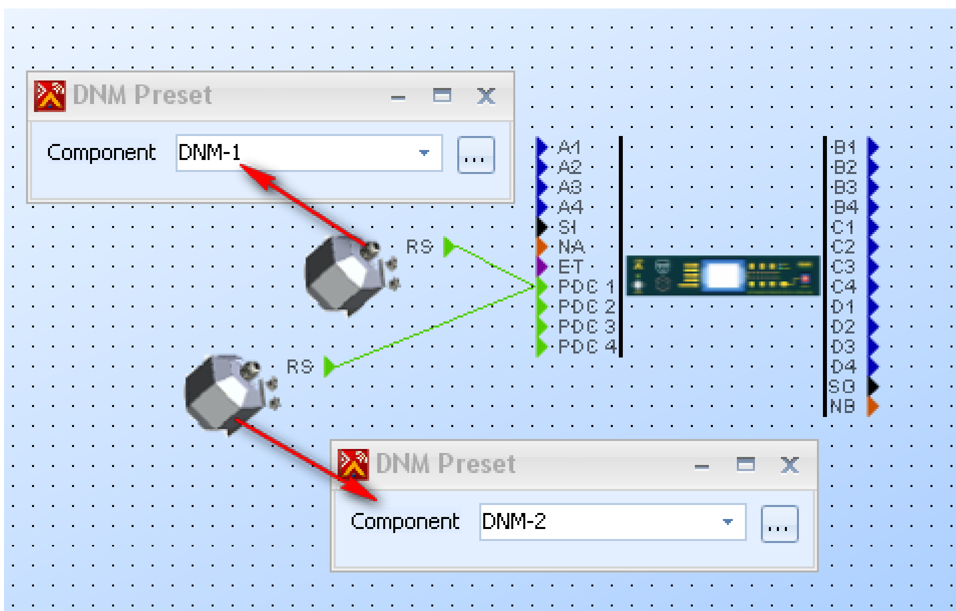
5. Draw DSP DNM component, and connect the input and output.



6. To modify the DSP component name. You can simply use property to do this.



7. Go to device editing window. Double click the device DNM component to open a window, and Link device DNM component with DSP DNM component.



8. Store, Online, Calibrate two DNMs by each. Please refer to preview topic "Configuration" for calibration procedure.

4.2.3 PPM AS

4.2.3.1 Overview

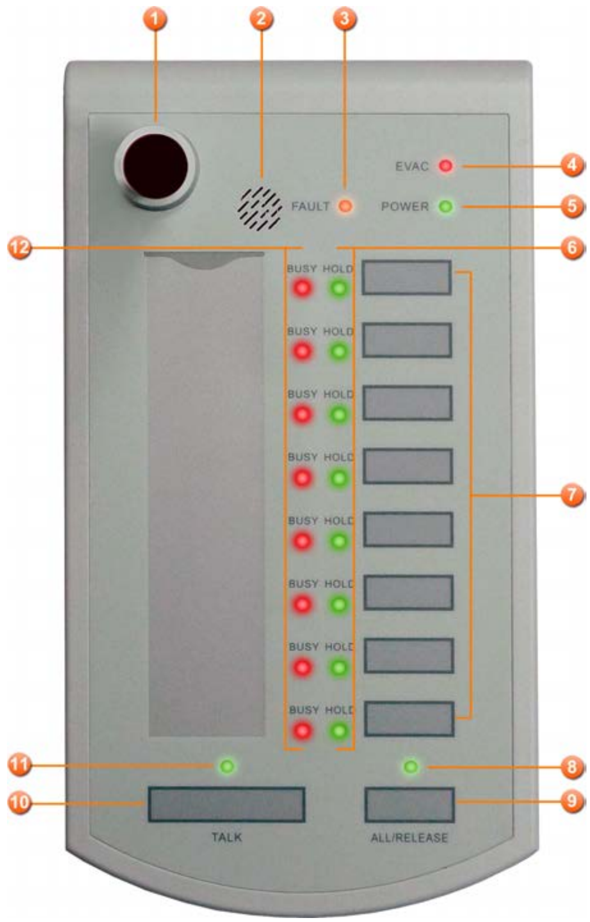
Unidirectional Condenser Addressable Microphone, compatible with all system units, PPM AS uses an RS485 protocol over a single CAT5 cable connection, to transport both Audio and Power from the PPM AS to the system units. The PPM AS comprises of 8 zones / 8 buttons with sleek condenser gooseneck microphone, and spring metal protection, providing durability and excellent aesthetics as well as allowing up to 256 zones expansion via the additional keypad easy extension station. The buttons can represent a single zone or a group of zones and are easily defined via the GUI of the system units using a simple Matrix selection.

The unit offers “Hold” and “Busy” LED signals in addition to the zone LED’s, and these allow the easy identification of selection / Busy signals for the user.

All buttons can be programmed with drag & drop features from the System unit GUI software and each button can be programmed for Push To Talk or Latch functionality.The unidirectional condenser microphone warrants high quality directive signal pick up from the user and hence less interference from the surroundings thanks to the cardioid polar pick-up pattern.

The RS485 communication protocol offers daisy chaining of up to 100 m on a CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connectors. (The microphone compatibility listing shows the maximum number of units per System).

4.2.3.2 Control Panel



1. Microphone

   The sleek condenser gooseneck microphone. This unidirectional condenser microphone warrants high quality directive signal pick up from the user and hence less interference from the surroundings.

2. Monitoring Speaker

   This embedded speaker to play chime when paging or monitor signal from audio processor, it needs to work with rs485 output component to get audio signal from other DSP components in audio processor.

3. Fault LED

   To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent	●	There are faults in audio processor.
Blinking	● ◦ ● ◦ ●	Lost communication with audio processor.

4. EVAC LED

Status	Frequency	Activity
Permanent	●	System is under evacuation paging.
Blinking	● ◦ ● ◦ ●	PPM AS does not do a "Plug-in" in Ateis Studio before using.

Plug-in" make audio processor to recognize the peripheral devices.

5. Power LED

   This LED light up when the PPM AS is powered on.

6. Hold LED

   These LEDs Indicate which keys are selected.

7. Event Select Key

   These buttons are used to select keys, each key may link to an event for triggering actions like paging or adjust the value of parameters.

8. All/Release LED

   This indicator lights up if all keys are selected.

9. All/Release Button

   This button is used to select or deselect all keys.

10. Talk Key

    Push this key to request paging.

11. Talk Indicator

    This LED light up if PCP is allowed to paging, on the other word, the paging request has been acknowledged.

    Sometime after paging request is granted, zones under paging still can be occupied by other sources. In such a case, the talk led still light up, but you can check busy LED to know the situation.

12. Busy LED

    Show the status of zone:

Color	Status	Frequency	Activity
Red	Permanent		Priority of the zone is lower than other sources.
Green	Blinking		Priority of the zone is higher than other sources.
Green	Permanent		The zone only desired for paging is not using by other sources.

4.2.3.3 Characteristics

❖ Case

Dimension = 105mm (W) x 190mm (L) x 50mm (H).

Weight = 0.7Kg.

Color = RAL7035.

Goose-neck length with microphone = 300 mm

❖ Power

Item	Voltage	Current Consumption	Comment

DC Input	18V~26V, Typical 24V	120mA	

❖Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 100Hz ~ 18kHz.

❖Front Panel Speaker

- Impedance = 4Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 8kHz.

❖Cable length

100m(control and power on the same cable)

900m(when power line connects to junction box).

❖Comes standard with Junction box (JB) and CAT5 cable (1.5)

4.2.3.4 Configuration

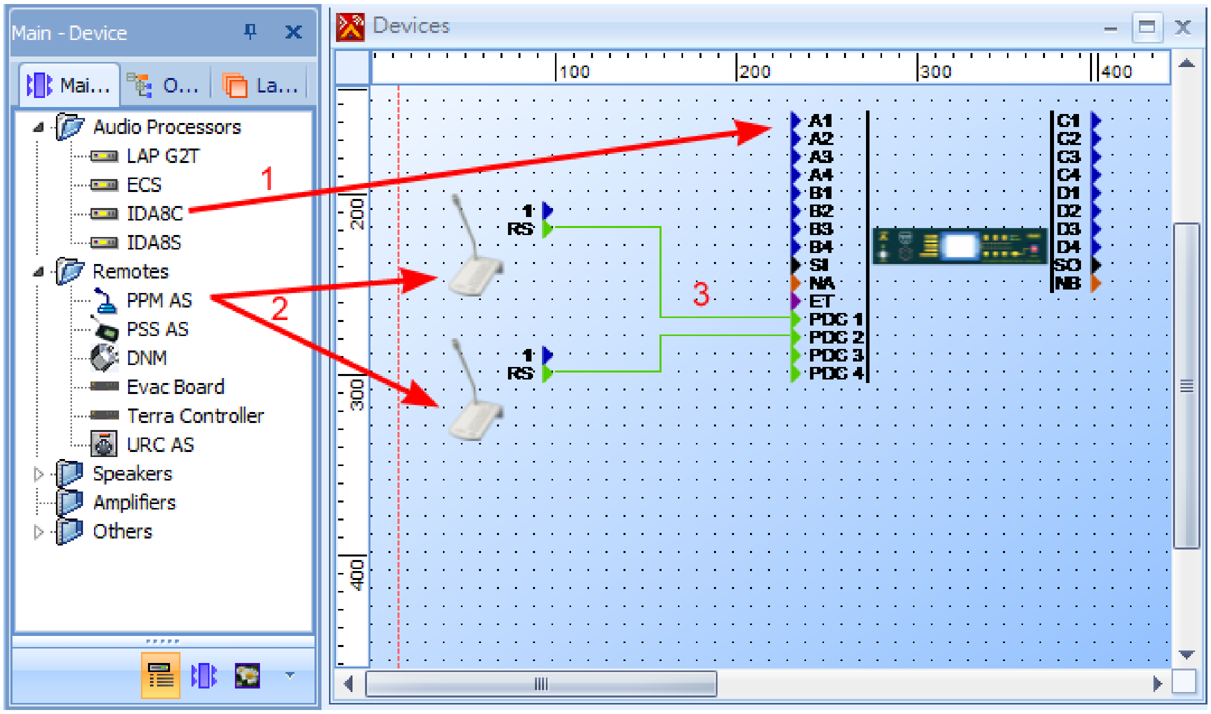
To make PPM AS work it need to edit configuration using Ateis Studio. Following sections are about configuration editing.

The example here is IDA8, the configuration is the same for ESC, LAPG2T and other Ateis Audio Processor.

❖Create PPM AS Object

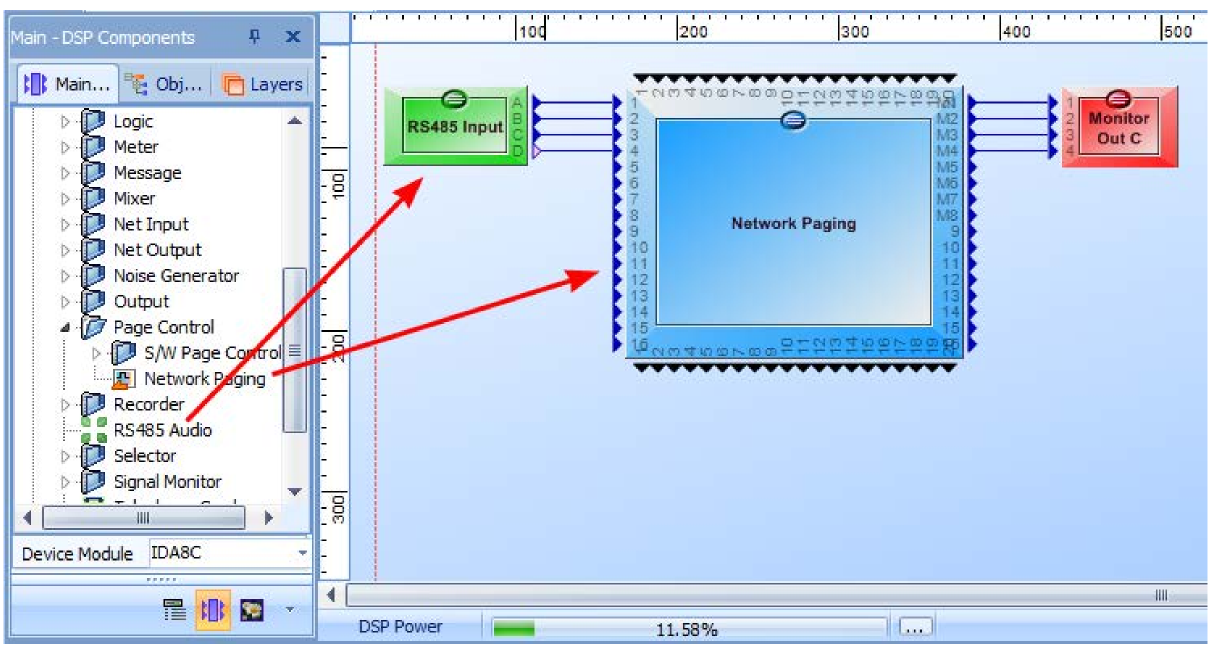
Open or create a Ateis studio configuration file, in the [Device] design window, create audio processor and PPM AS object:

1. Drag IDA8C to design window to create audio processor object.
2. Drag PPM AS to design window to create PPM AS object.
3. Wiring to connect green pins between PPM AS object and audio processor.



❖ DSP component configuration

It need to route audio from PPM AS to paging component for paging processing. See the following example:

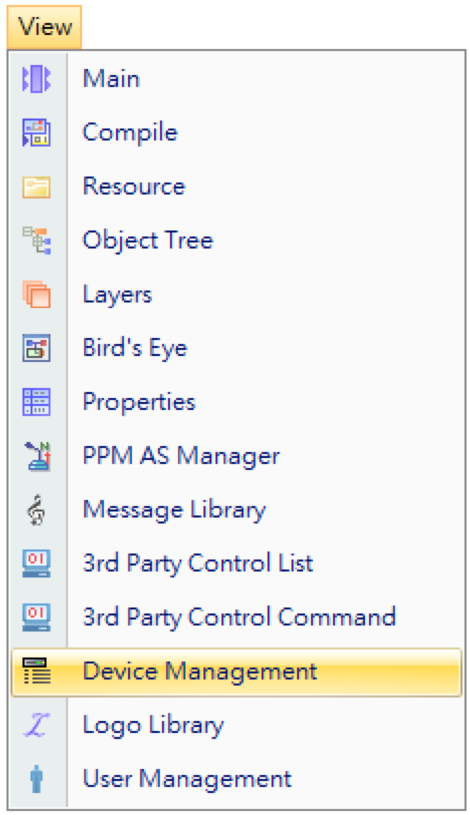


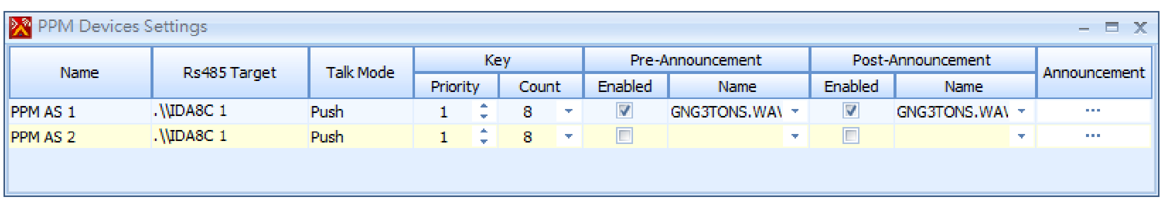
Pin A~D of RS485 Input component is corresponds to PPM AS 1~4 audio plugged in PDC1~4 respectively. Network Paging component process paging request coming from various sources, and to determine which source is allowed to paging to zones it desired. The zones are M1~M8 on Network Paging component.You need create an integration paging event to link with PPM AS, See topic Network Paging to learn editing paging settings.

❖ PPM AS Manager

PPM AS Manager is an unit of the software to manage settings about PPM AS that connect to audio processors like IDA8 series.

Click main menu of software [View > PPM AS Manager] to open PPM AS Manager:





Name	Rs485 Target	Talk Mode	Key		Pre-Announcement		Post-Announcement		Announcement
			Priority	Count	Enabled	Name	Enabled	Name	
PPM AS 1	.\\IDA8C 1	Push	1	8	☑	GNG3TONS.WA\	☑	GNG3TONS.WA\	...
PPM AS 2	.\\IDA8C 1	Push	1	8	☐		☐		...

- Name

  The name of PPM AS object.

- Rs485 Target

  Indicates the connector on audio processor that PPM AS connected in the design window.

- Talk Mode(todo rider)

  To specify the behavior of key pressing for paging. There are two modes:

  Push: Push and hold the key for paging request, release the key to end paging.

  Lock: First push for paging request, if request is grant then start paging, push again to end paging.

- Key-Priority(todo: rider)

  The priority of key.

- Key-Count

  To specify how many keys on PPM AS.

- Pre-Announcement-Enabled

  Enable or disable pre-announcement chime playing.

- Pre-Announcement-Name

  The name of pre-announcement message.

- Post-Announcement-Enabled

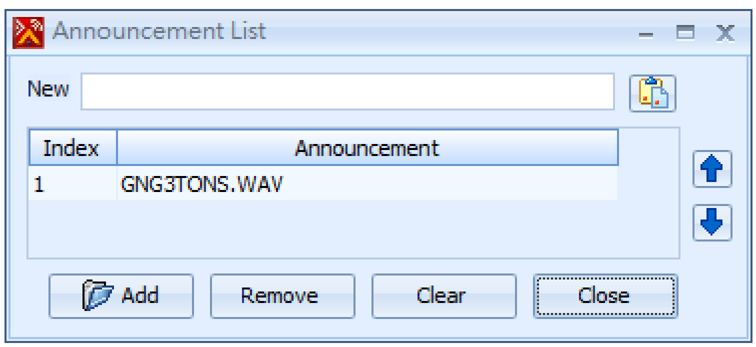
  Enable or disable post-announcement chime playing.

- Post-Announcement-Name

  The name of post-announcement message.

- Announcement

  Open announcement message settings window:



  - New

    Input a name of announcement message for creating new entry in the list. Type name in the editor and press the button right to the editor.

  - Index

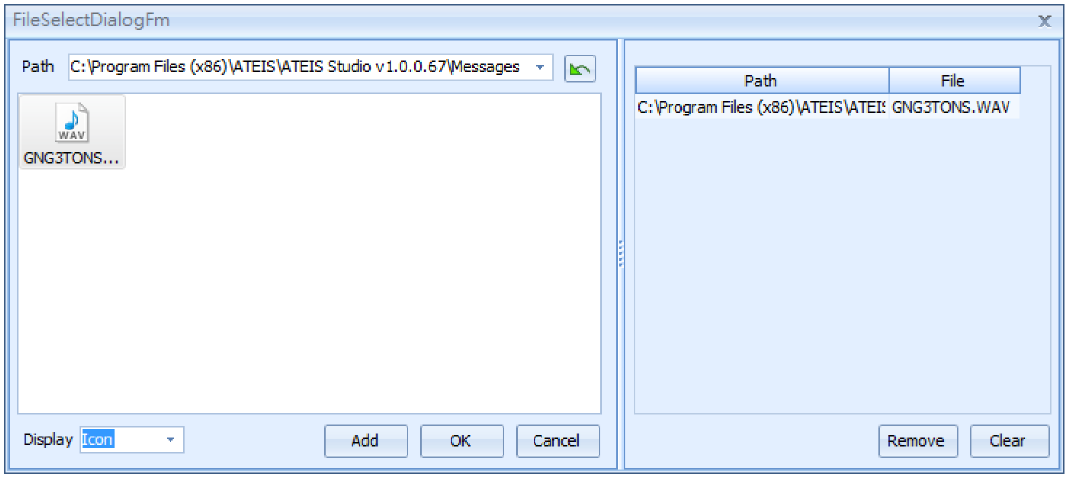
    Index for entries in the list.

  - Announcement

    The name of announcement message.

  - Add

    Add a name of announcement message from File Selection Dialog:



- Path

  Specify the directory to list all files with supported formats.

- File List

  Display files in the directory Path.

- Display

  Specify the display mode for file list. There are three modes: Icon/List/Detail.

- Add

  Move the select files to the ready list that the right side of window.

- Ok

  Add files in the ready list to announcement message list.

- Cancel

  Close window and ignore ready list.

- Path

  The directory name of message file belongs to.

- File

  The name of message file.

- Remove

  Remove the select message file in ready list.

- Clear

  Clear ready list.

o Remove

Remove a message from list.

o Clear

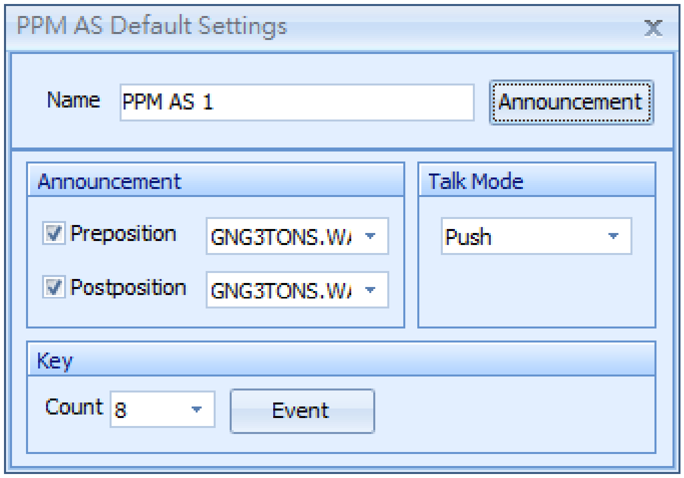
Clear the list.

- Close

Close the window.

❖ PPM AS Settings

Double click the PPM AS object open the window of PPM AS Settings:



- Name

  The name of PPM AS.

- Announcement

  Open announcement message settings window, see prior section for details.

- Announcement Preposition

  Enable or disable pre-announcement chime playing.

- Announcement Preposition Name

  Select message from the announcement message list for pre-announcement chime playing.

- Announcement Postposition

  Enable or disable post-announcement chime playing.

- Announcement Postposition Name

  Select message from the announcement message list for post-announcement chime playing.

- Talk Mode

  Specify the behavior of key pressing when paging. See prior section for details.

- Key Count

  Specify how many number of keys in PPM AS.

- Key Event

  Bind a paging event with PPM AS for zone selection. See Network Paging component topic for

more details.

❖ Remote Plug-in

It need to do a "Remote Plug-in" to make audio processor recognize this remote devices. See the topic Remote Plug-in for more details.

❖ Assign to Design Objects

The device object in design window needs map to physical device, then the configuration of the device can be set to the physical one. Click "Name " field of device list to select correspond device object. Refer to topic search and settings for more detail.

4.2.4 PSS AS

4.2.4.1 Overview

The PSS AS touch paging microphone console is a man-machine interface which allows paging call, messages broadcasting and DSP matrix parameter control. Its back-lit touch screen is designed for simple and user-friendly operating. Various operating levels with password protection make the PSS AS a versatile device that fits well in a commercial shopping center as for an industrial high security environment.

All paging parameters needed for site operating can be programmed: zones assigned to the different buttons, name of zones, group of zones, messages triggering, levels adjustments and pre-call chime but also for fader control, button control or event control.

A total of twelve keys on fourteen pages allow zone or group of zones selections. Each key contains a color changing field indicating that the zone is occupied by a different process.

4.2.4.2 Control Panel

1. Microphone

    A 280mm flex microphone.

2. Power LED

    This LED light up when the PSS AS is powered on.

3. Fault LED

    To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent		There are faults in audio processor.
Blinking		Lost communication with audio processor.

4. EVAC LED

    Light up when system is under evacuation paging.

5. Touch Screen

A back-lit touch screen with 5" diagonal and 800 x 480 resolution contains 14 pages, 12 keys for each pages can freely config to paging or adjustment parameters in audio processor.

6. Left Button

   No function for the moment.

7. Middle Button

   No function for the moment.

8. Right Button

   Press and hold it for a while to calibrate touch screen.

9. Monitoring Speaker

   This embedded speaker to play chime when paging or monitor signal from audio processor, it needs to work with rs485 output component to get audio signal from other DSP components in audio processor.

4.2.4.3 Characteristics

❖Case

Dimension = 250mm (W) x 140mm (L) x 75mm (H).

Weight = 1.1Kg.

Color = RAL7016.

Goose-neck length with microphone = 300 mm

❖Screen

Diagonal = 5" .

Resolution = 800 x 480.

❖Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	250mA	-

❖Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 100Hz ~ 18kHz.
- Noise Gate Threshold = -84dBu ~ -24dBu.
- Target Output Level = 0.
- Max Output Level = -54dBu ~ 6dBu.
- Attack Time = 8ms.
- Release Time = 100ms.

❖Front Panel Speaker

- Impedance = 4Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 12kHz.

❖Maximum Cable Length

100m on Category 5 cable

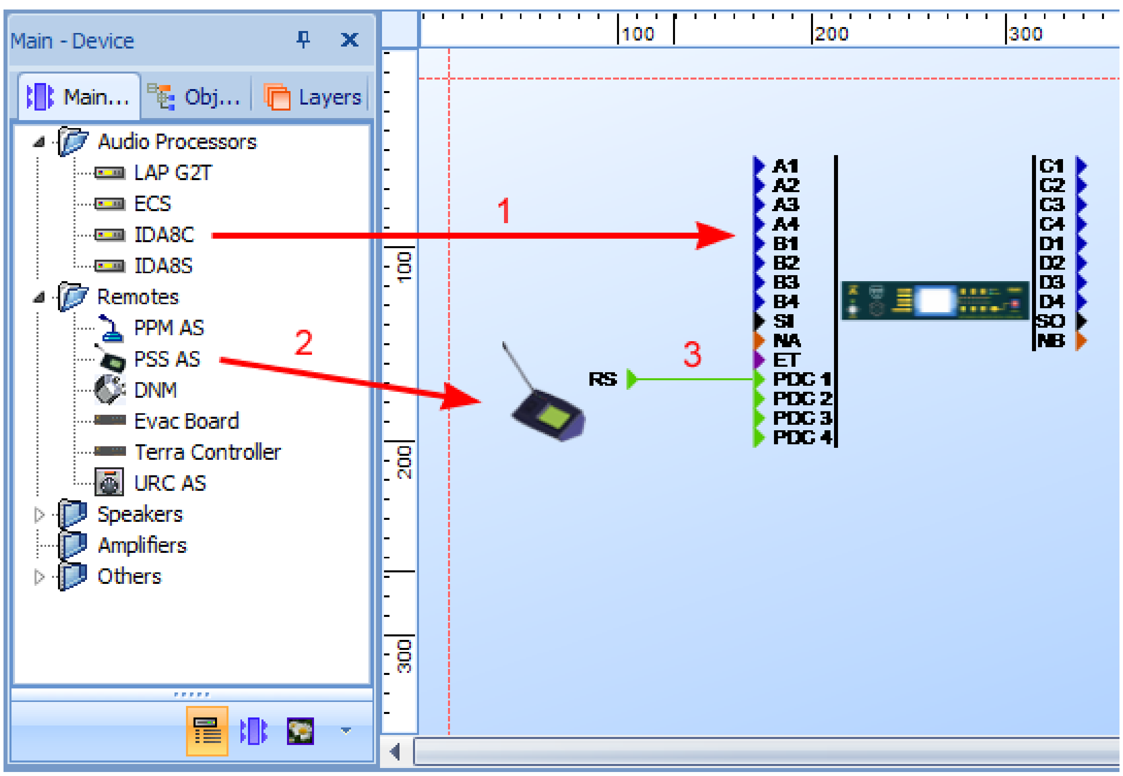
4.2.4.4 Configuration

To make PPM AS work it need to edit configuration using Ateis Studio. Following sections are about configuration editing.

❖Create PPM AS Object

Open or create a Ateis studio configuration file, in the [Device] design window, create audio processor and PSS AS object:

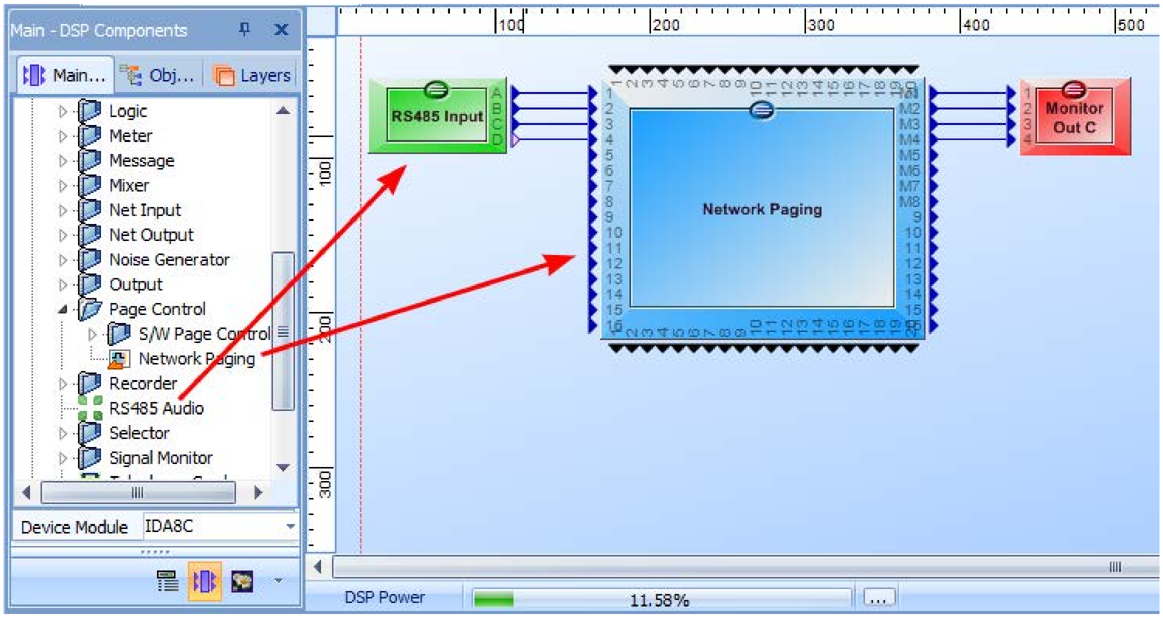
1. Drag audio processor to design window to create audio processor object.
2. Drag PSS AS to design window to create PPM AS object.
3. Wiring to connect green pins between PSS AS object and audio processor.



❖DSP component configuration

It need to route audio from PPM AS to paging component for paging processing. See the following

example:



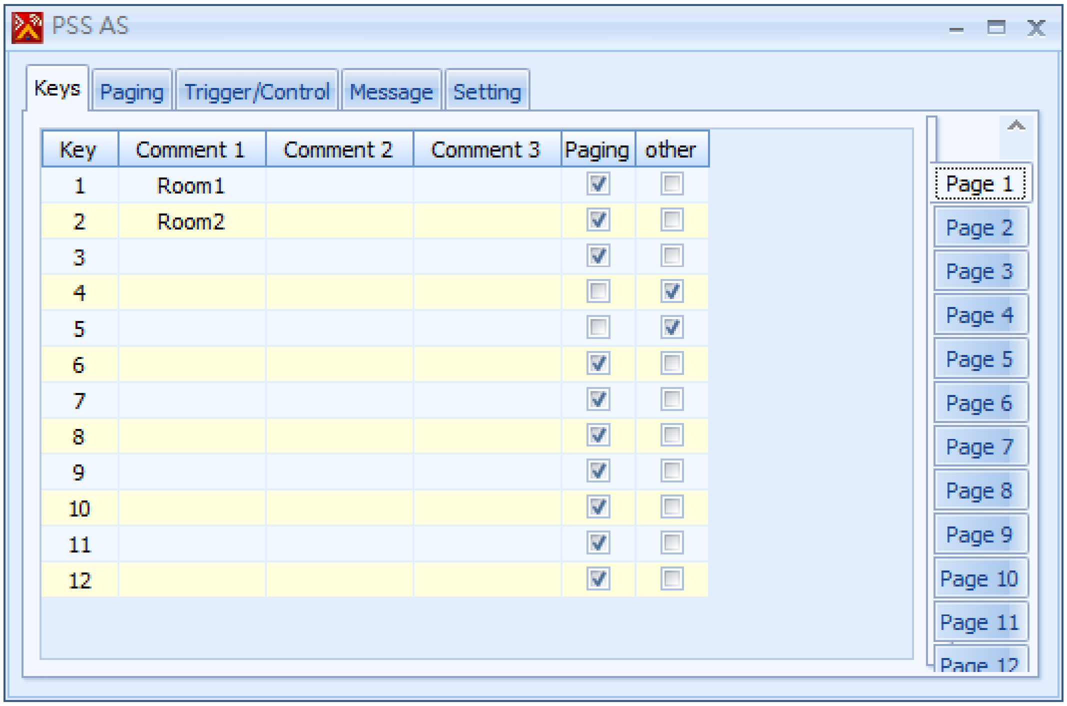
Pin A~D of RS485 Input component is corresponds to PPM AS 1~4 audio plugged in PDC1~4 respectively. Network Paging component process paging request coming from various sources, and to determine which source is allowed to paging to zones it desired. The zones are M1~M8 on Network Paging component. You need create an integration paging event to link with PSS AS, see topic Network Paging to learn editing paging settings.

❖PSS AS Settings

Double click PSS AS object to open settings window, There are 14 pages on the right of window, each or page is corresponds to a page on PSS AS. There are five tabs on the top of window containing various settings of each. The following sections describe for each settings of tab.

❖PSS AS Settings-Key Tab

This page is the settings about the captions for each key and the usage selection.



- Key

  Refer to the key on the page of PPM AS.

- Comment 1/2/3

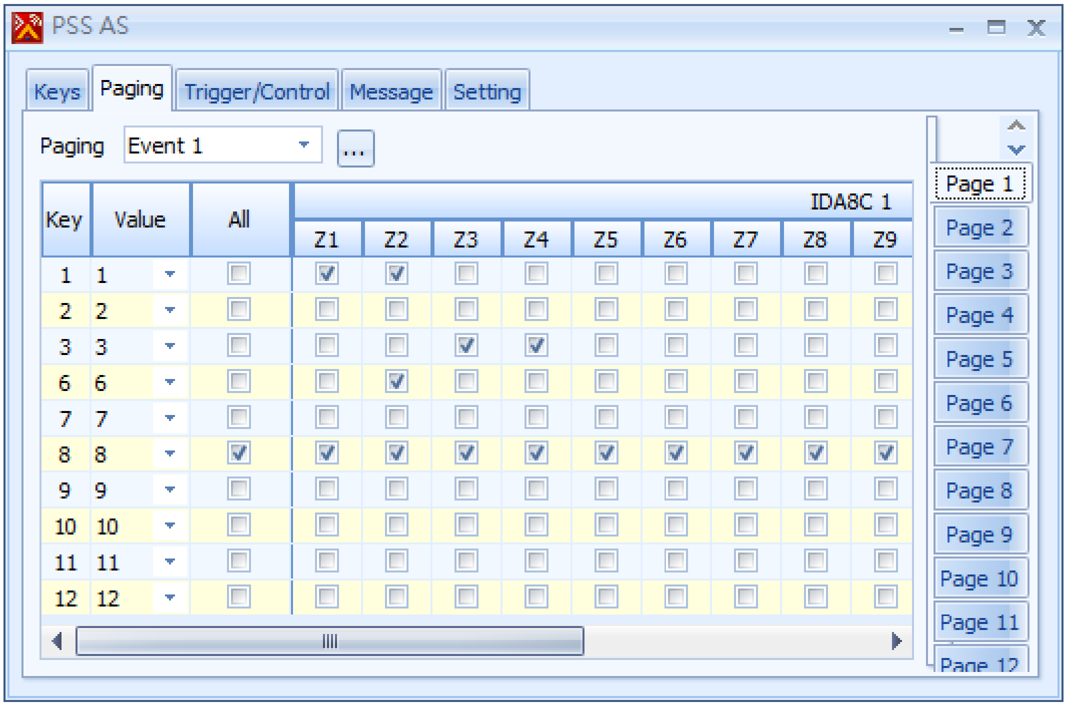
  The text displayed on the key, there are three rows on it Comment 1 is the first row and so on.

- Paging/Other

  Check boxes for Paging and Others option are alternative. Choose one of them to determine the key used for paging or triggering an event.

❖PSS AS Settings-Paging Tab

This page shows the zone combination for each key which is configured as paging.



- Value

  To specify which code on integration paging event is bound with the key.

- All

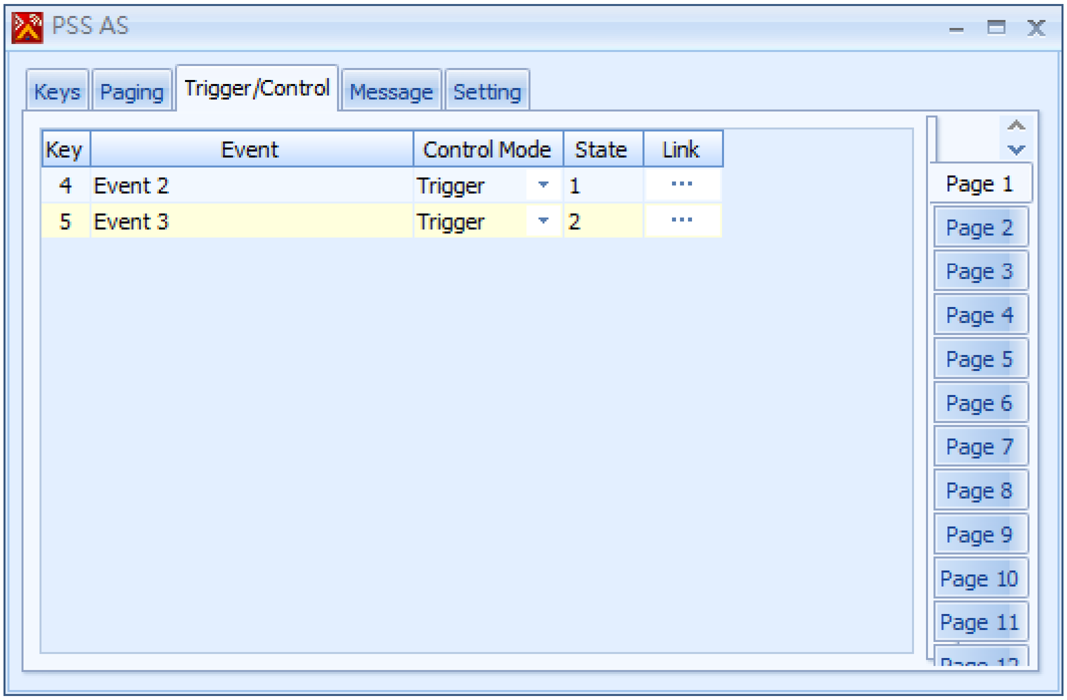
  Select all zone for the key.

- Zone Selection

  The grid list all zones of network paging component for all audio processors. click check boxes to select zones separately for the key.

❖PSS AS Settings-Trigger/Control Tab

This section mainly about settings of event linking. PSS AS with the ability of triggering event using key. You need to specify the link relation of key and event at design time.



- Event

  Select the event linked to.the key.

- Control Mode

  Specify the control mode is Trigger or Control, see more details on topic Event Management.

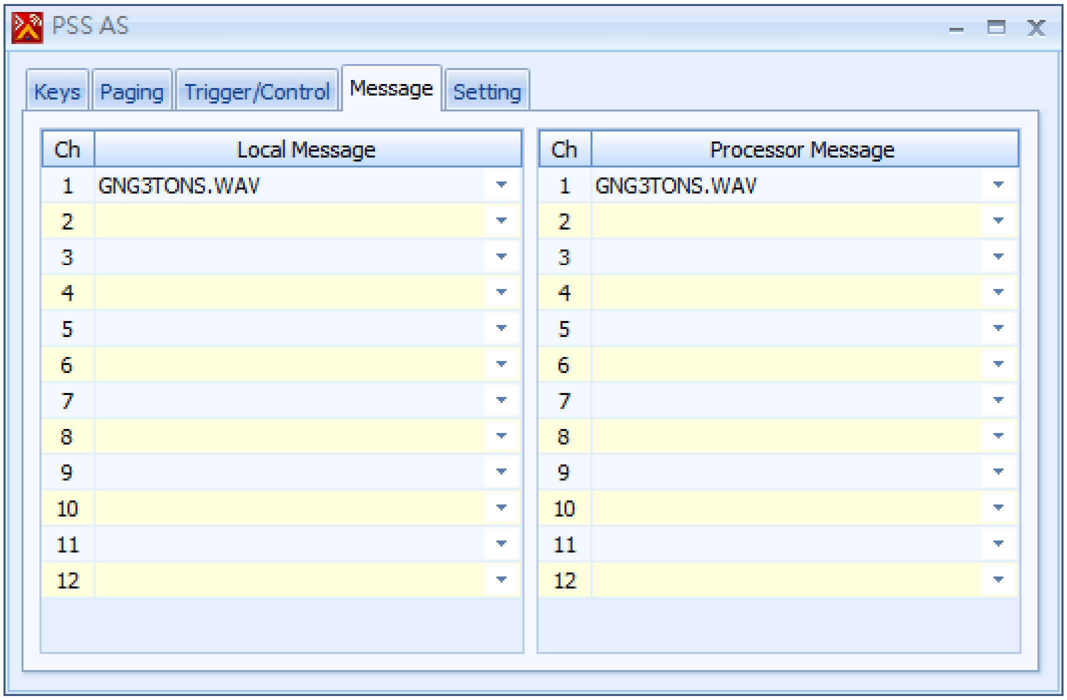
- State

  Enter a value for the state that processed by event control procedure, each state number result different adjustment of system, see more details on topic Event Management.

- Link

  Open settings window of the event.

❖PSS AS Settings-Message Tab



This section is talking about to setup the message list for play on runtime in PSS AS. Before selection, you need to build chime list which settings in on the tab Setting.(todo check with rider)

- Ch(Left)

  The channel number for a message on local.

- Local Message

  The message belongs to local list, indicates this message is inside the PSS AS itself.

- Ch(Right)

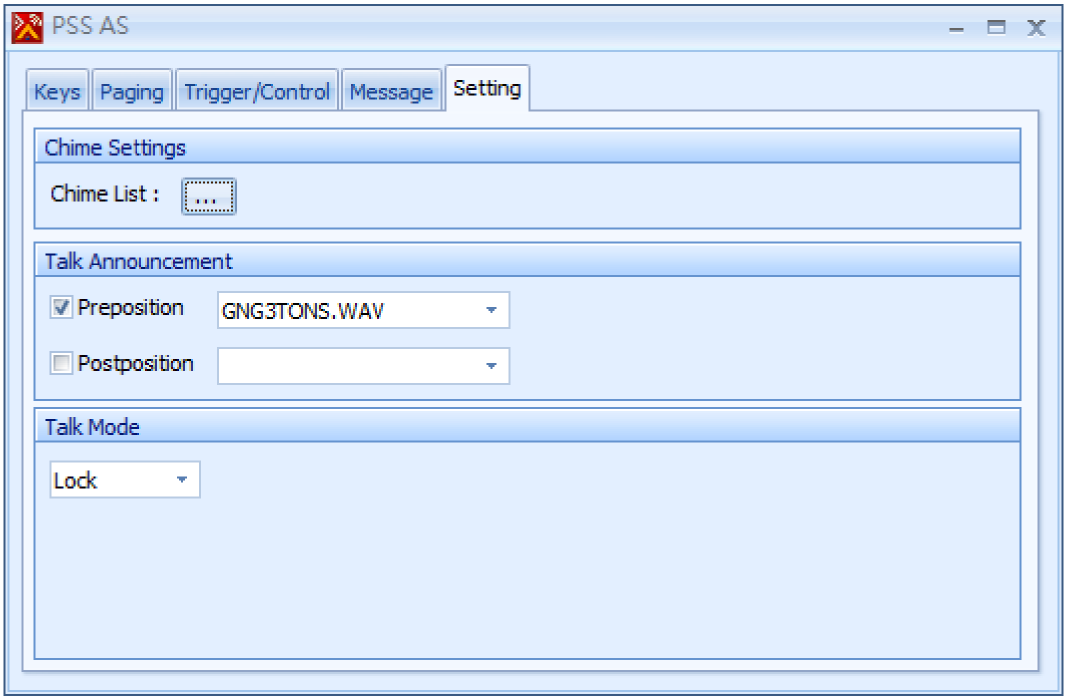
  The channel number for a message on audio processor.

- Processor Message

  The message belongs to processor list, indicates this message is inside the audio processor.

  ⚠ It need to work with Message Player component.

❖ PSS AS Settings-Setting Tab



This section is settings about chimes.

- Chime List

  Open a settings window to build up list of message for selection.

- Pre-announcement Enable

  To enable or disable pre-announcement.

- Pre-announcement Chime Selection

  Select a message for pre-announcement.

- Post-announcement Enable

  Enable or disable post-announcement.

- Post-announcement Chime Selection

  Selection a message for post-announcement.

- Talk Mode

  There are two mode available:

  Push: Push and hold the key for paging request, release the key to end paging.

  Lock: First push for paging request, if request is grant then start paging, push again to end paging.

❖ Remote Plug-in

It need to do a "Remote Plug-in" to make audio processor recognize this remote devices. See the topic Remote Plug-in for more details.

❖ Assign to Design Objects

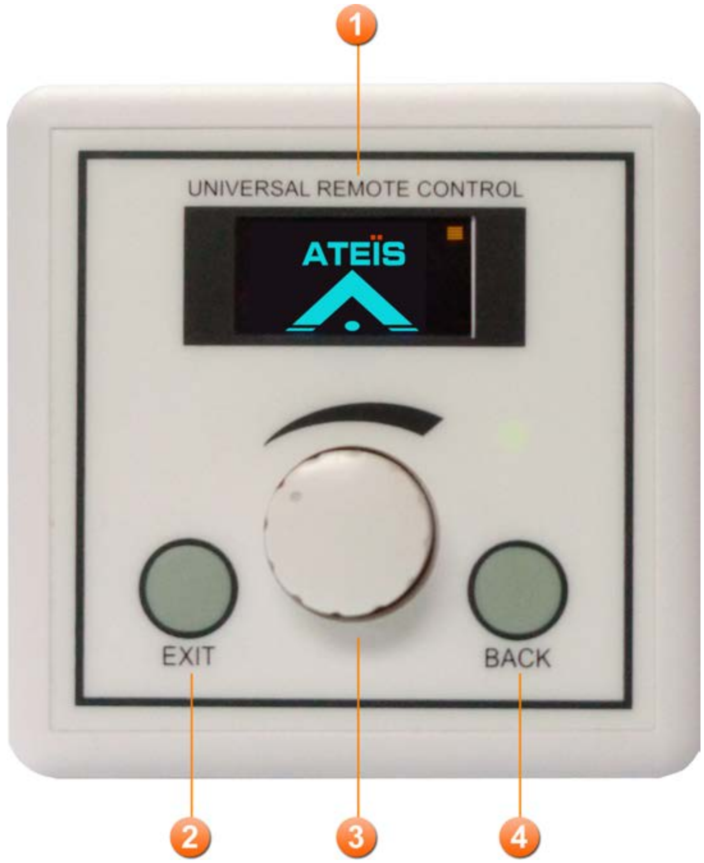
The device object in design window needs map to physical device, then the configuration of the device can be set to the physical one. Click "Name " field of device list to select correspond device object. Refer to topic search and settings for more detail.

4.2.5 URC AS

4.2.5.1 Overview

The URC AS can be fully programmed via Ateis Studio to adjust every setting you want: volume, mute, preset, components' adjustments... URC AS connect to audio processor via RS485 cable that allow you to remote control parameters of audio processor with a long distance. A elegant OLED to display information of parameters or the status. It give extreme simple and effective experience of control interface with only Two buttons [EXIT], [BACK] and a Knob to achieve whatever action desired. The URC AS is definitely the best choose for remote controlling with low cost consideration.

4.2.5.2 Control Panel



1. OLED Display
2. Exit Button

   Back to the main menu.
3. Knob

   Turn: Select items or change volume.

   Press: Enter in the sub menu.
4. Back Button

   Back to the root menu.

4.2.5.3 Characteristics

❖ Case

Dimension = 84mm (W) x 33mm (L) x 84mm (H).

Weight = 0.08KG.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	24VDC	20mA	

❖Maximum Cable Length

1200m on Category 5 cable

4.2.5.4 Configuration

To make URC AS work it need to edit configuration using Ateis Studio. Following sections are about configuration editing.

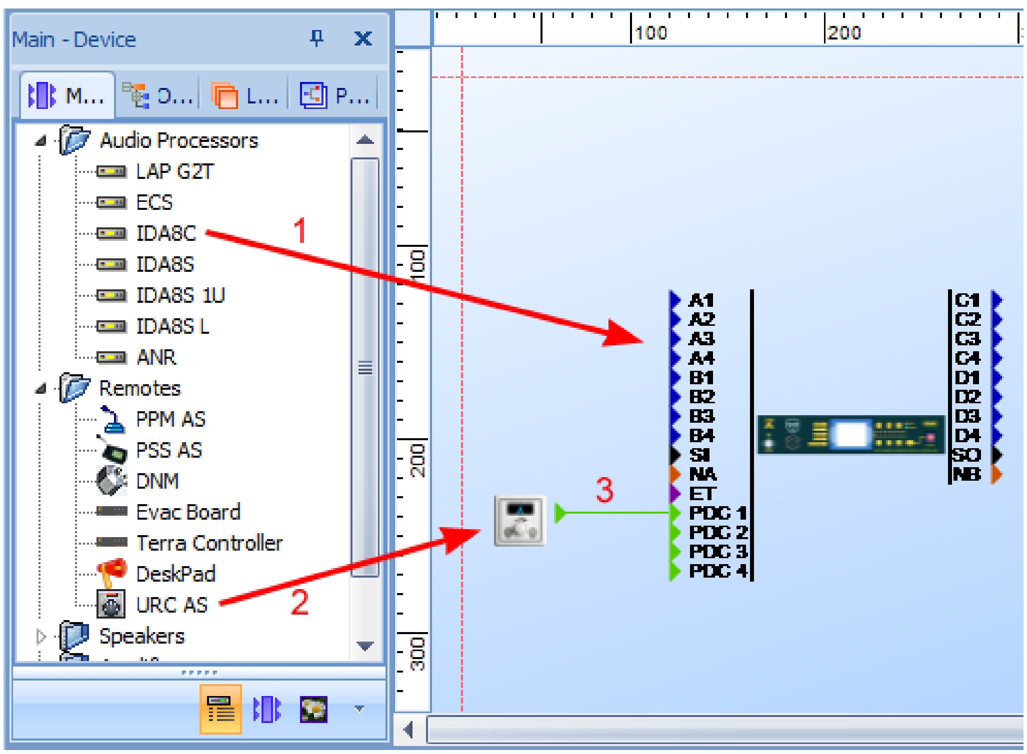
❖Create URC AS Object

Open or create a Ateis studio configuration file, in the [Device] design window, create audio processor and URC AS object:

1. Drag audio processor to design window to create audio processor object.

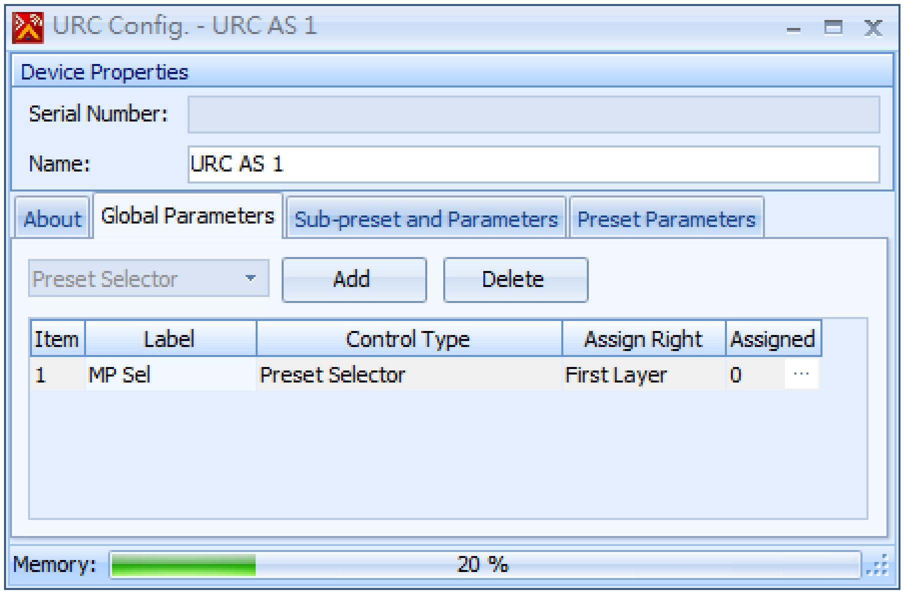
2. Drag URC AS to design window to create URC AS object.

3. Wiring to connect green pins between URC AS object and audio processor.



❖URC AS Settings

Double click URC AS object to open settings window.



- Serial Number

  Serial number of the URC AS.

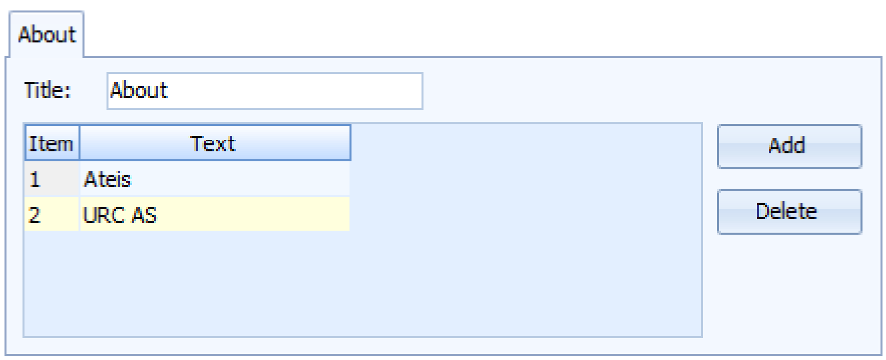
- Name

  Name of the URC AS.

- Memory

  Indicate memory usage of the URC AS for current settings.

❖URC AS Settings-About Tab



This page include settings for About function in URC AS. About is the last item of first layered menu in URC AS. Click the item to enter second layer of About function that is a list of texts that set in design time.f

- Title

  The text of about title.

- Item

  The item No. for the text in About text list.

- Text

  The text in About text list.

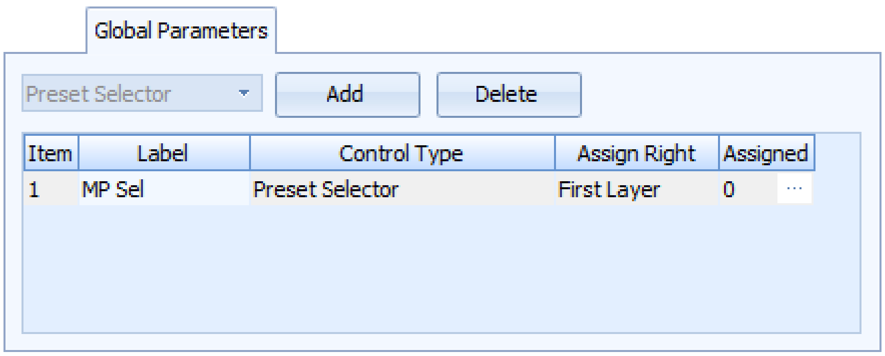
- Add

  To add a text in About text list.

- Delete

  To remove selected text from About text list.

❖URC AS Settings-Global Parameters Tab



This page is settings about items in URC AS for controlling global parameters like changing master/ sub presets. Users are allow to create arbitrary number of items until reach the memory limit which displayed on the bottom of window.

- Add

  Add a item in URC AS for controlling global parameters.

- Delete

  To remove a item.

- Item

  Index of the item.

- Label

  Display text for the item.

- Control Type

  To specify which type of global parameter is going to control. Available types list below:

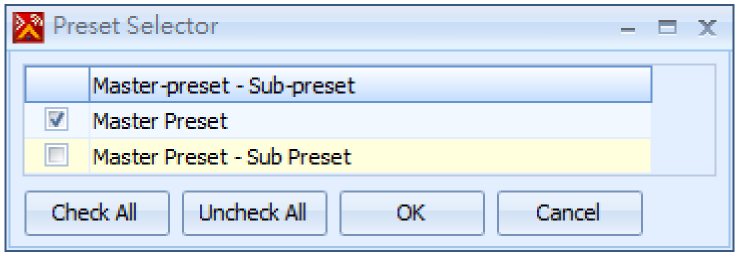
  - Preset Selector : To change master/sub presets.

- Assign Right

  To specify the item is belong to first layer or protected layer.

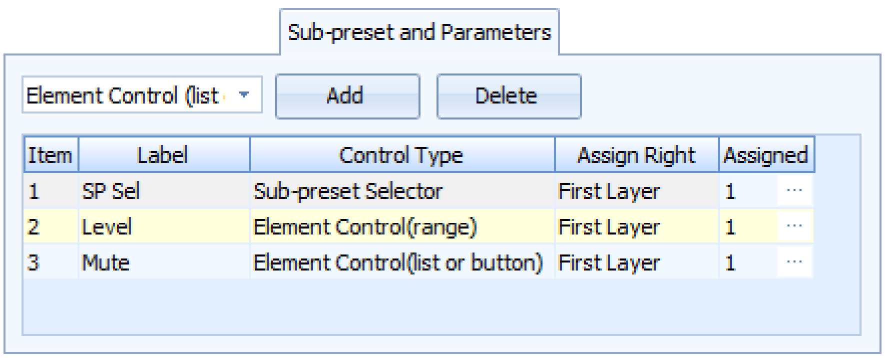
- Assigned

  Indicate the number of presets are selected for change. There is an ellipsis button on the right of grid cell, click it to open the window for preset selection.



- Check boxes for selection of master/sub presets to be linked to the item.
- Check All button to select all master/sub presets.
- Uncheck All button to de-select all master/sub presets.
- OK button to confirm settings and close window.
- Cancel to discard the modification of settings and close window.

❖ URC AS Settings-Sub-preset and Parameters Tab



This page is settings about sub-preset and parameter control using items in URC AS.

- Function Combo Box

  To specify which function for controlling going to be created. The available functions list below:

  - Sub-Preset Selector

    To change sub-preset.

  - Element Control(Range)

    To control element with range type value. For example "Level" in Input component with min -90 dB and max 20 dB is a range type of element.

  - Element Control(List or Button)

    To control element with list or boolean type value. For example "Mute" in Input component with states On/Off is a button type of element.

- Add

  To add an item for controlling sub-presets or elements.

- Delete

  To remove selected items in the list.

- Item

  Index of the item.

- Label

  The display text of the item.

- Control Type

  The type of item. refer to prior section "Function Combo Box" for more details.

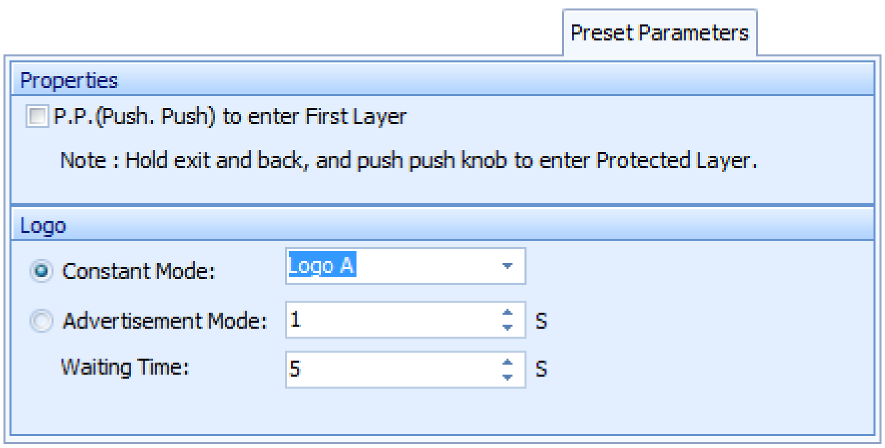
- Assign Right

  To specify the item is belong to first layer or protected layer.

- Assigned

  Indicate how many controlled target linked to the item, for example if a item controls 4 channels of "Mute" simultaneously, the value "Assigned" will be 4.

❖ URC AS Settings-Preset Parameters Tab



This page is settings about behavior of menu changing.

- P.P. to enter First Layer

  If this option is enabled, Double clicks to enter first layer menu is required.

- Constant Mode

  There are two picture (Logo A/Logo B)stored in an URC AS for logo. Under this mode the logo is display permanently using the one selected on combo box.

- Advertisement Mode

  In this mode two pictures are display in turn, the time to switch picture is in the editor on the right.

- Waiting Time

  To specify how long will show Logo pictures when URC AS idled.

❖ Remote Plug-in

It need to do a "Remote Plug-in" to make audio processor recognize this remote devices. See the

topic Remote Plug-in for more details.

❖ Assign to Design Objects

The device object in design window needs map to physical device, then the configuration of the device can be set to the physical one. Click "Name " field of device list to select correspond device object. Refer to topic search and settings for more detail.

4.2.6 URGP

4.2.6.1 Overview

The URGP is an extension unit to the IDA8-Systems. Each URGP will provide those systems with 32 additional alarm inputs. Each input is monitored and can be programmed to trigger a digital audio message into a specific zone or group of zones. The URGP is linked to the System units trough a RS232 /RS485 monitored serial link.

4.2.6.2 Characteristics

❖ Case

Dimension = 140mm (W) x 75mm (L) x 44mm (H).

Weight = 0.48KG.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	24VDC	30mA	-

❖ Maximum Cable Length

RS485 = 600m on Category 5 cable

RS232 = 15m

❖ Evacuation Inputs(Contact Mode)

- Bias voltage:

Item	Minimum	Maximum	Unit
Voltage	-	5	VDC

- Monitoring resistor = 4.7k Ohm.

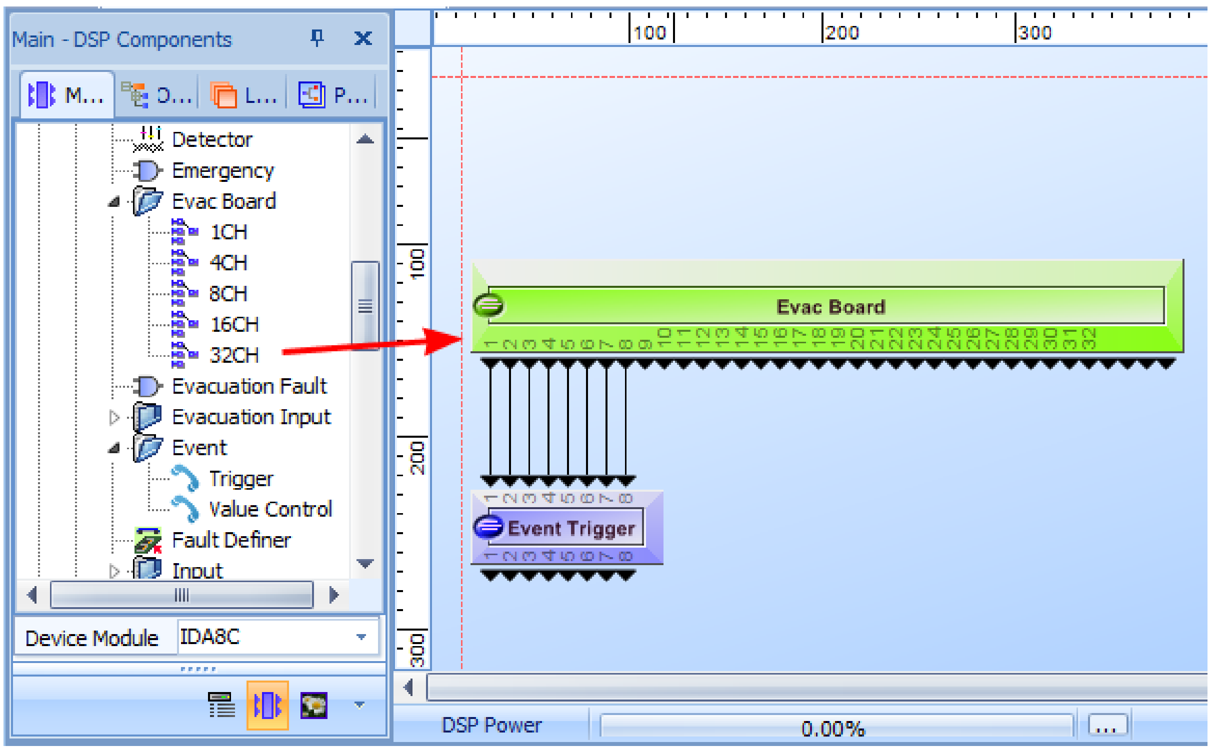
❖ Evacuation Inputs(Voltage Mode)

On Voltage:

Item	Minimum	Maximum	Unit
Voltage	18	72	VDC

4.2.6.3 Configuration

URGP needs config using Ateis Studio before working. EVAC Board is the DSP component for the device.



See more details topic EVAC Board.

4.2.7 CD-Touch

4.2.7.1 Overview

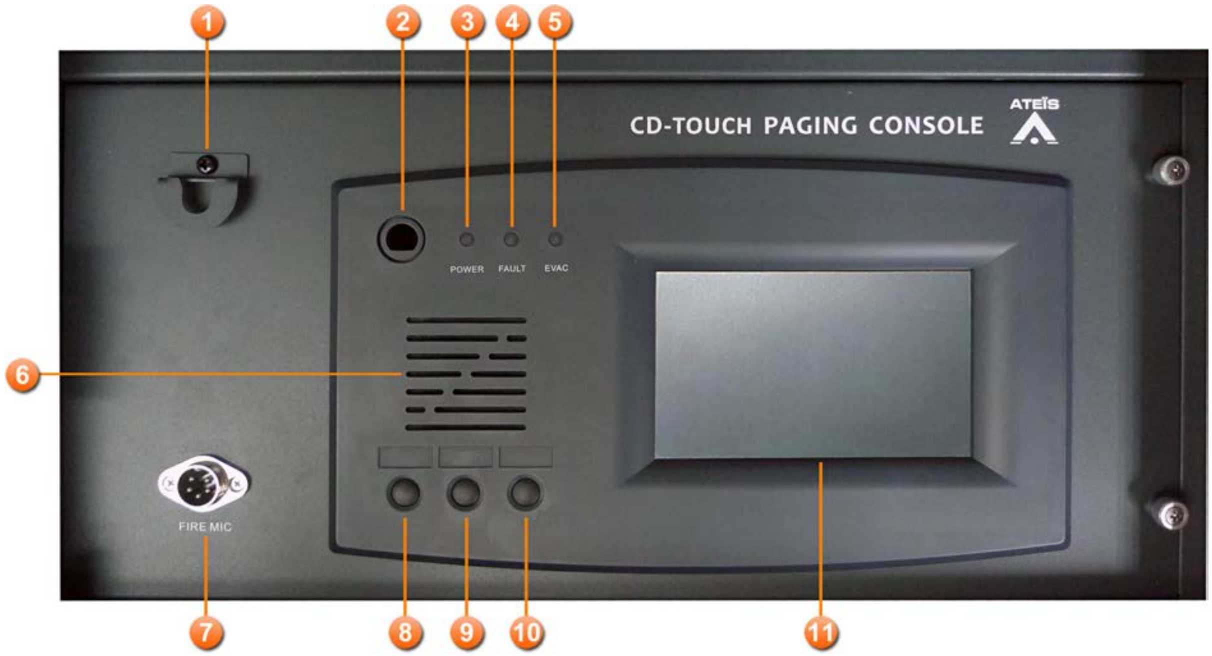
CD-Touch Wall mounted cabinet remote paging console for Ateis audio processor.

Wall-mount heavy duty remote paging console with Access level 2 protection to comply with the EN 54-16. Compatible with Ateis audio processor and communicates over a dedicated RS485 for Power, audio and DATA.

The unit comprises of 8 zones / 8 buttons with fist firemen's microphone in a metal surface mount wall-box. It provides robust IP-30 protection.

The RS485 communication protocol offers daisy chaining of up to 300 m on a simple CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connector.

4.2.7.2 Control Panel



1. Fireman Microphone Hook

   An U-shape hook to place fireman microphone.

2. Microphone

   Not used.

3. Power Indicator

   This LED light up when the CD-Touch is powered on.

4. Fault Indicator

   To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent		There are faults in audio processor.
Blinking	∘ ∘	Lost communication with audio processor.

5. Evacuation Indicator

   Light up when system is under evacuation paging.

6. Monitoring Speaker

   This embedded speaker to play chime when paging or monitor signal from audio processor, it needs to work with rs485 output component to get audio signal from other DSP components in audio processor.

7. Fireman Microphone Connector

   DIN Connector for fireman microphone connection.

8. Left Button

No function for the moment.

9. Middle Button

No function for the moment.

10. Right Button

No function for the moment.

11. Touch Screen

A back-lit touch screen with 5" diagonal and 800 x 480 resolution contains 14 pages, 12 keys for each pages can freely config to paging or adjustment parameters in audio processor.

4.2.7.3 Characteristics

❖ Case

Dimension = 400mm (W) x 210mm (H) x 130mm (D).

Weight = 4.5Kg.

Color = RAL7035.

❖ Screen

Diagonal = 5" .

Resolution = 800 x 480.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	250mA	

❖ Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 0.5%.
- Bandwidth @ -3dB = 50Hz ~ 20kHz.
- Noise Gate Threshold = -84dBu ~ -24dBu.
- Target Output Level = 0.
- Max Output Level = -54dBu ~ 6dBu.
- Attack Time = 8ms.
- Release Time = 100ms.

❖ Front Panel Speaker

- Impedance = 4Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 0.5%.

- Bandwidth @ -3dB = 50Hz ~ 20kHz.

❖Maximum Cable Length

100m on Category 5 cable

4.2.8 CD8

4.2.8.1 Overview

CD8 Wall mounted cabinet remote paging console for Ateis audio processor.

Wall-mount heavy duty remote paging console with Access level 2 protection to comply with the EN 54-16. Compatible with Ateis audio processor and communicates over a dedicated RS485 for Power, audio and DATA.

The unit comprises of 8 zones / 8 buttons with fist firemen’s microphone in a metal surface mount wall-box. It provides robust IP-30 protection. Each CD8 contains a PMM PS Master PCB with extension keypad and uses the same architecture as for the PPM AS series of microphone consoles. Each Ateis audio processor like IDA8C can handle up to 31 CD8 units per 485 port in Master/Slave configuration. (Note that only the Master unit is secured, Slaves are not).

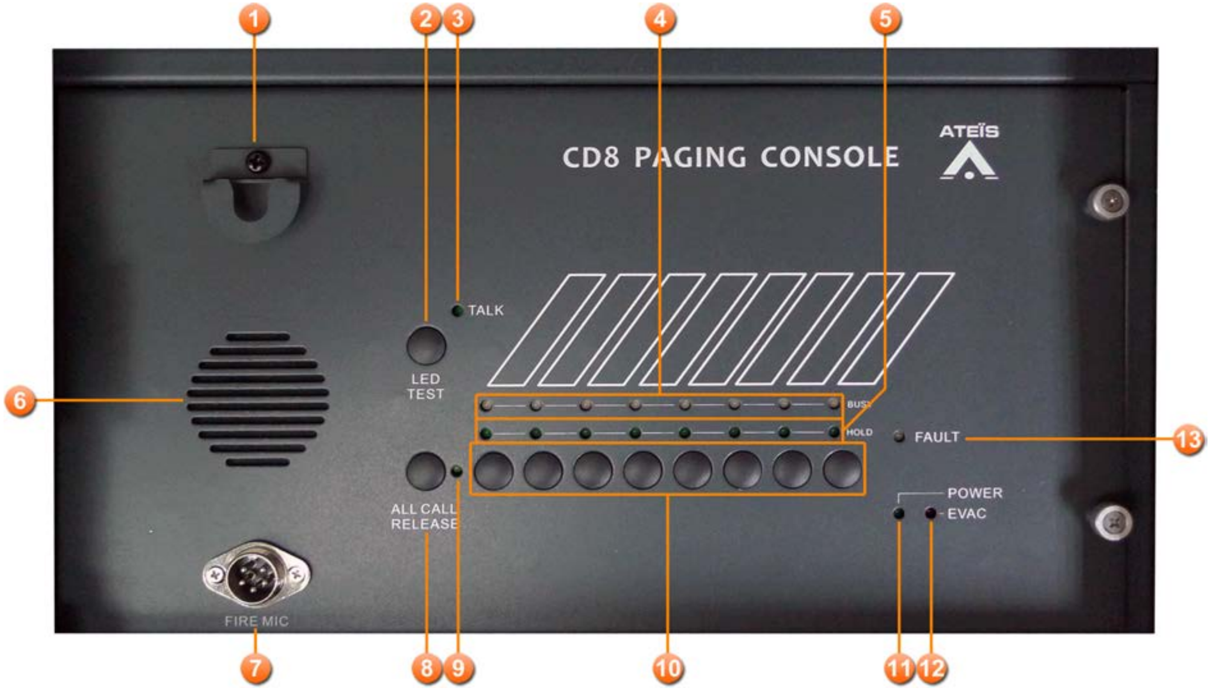
The buttons can represent a single zone or a group of zones. All buttons can be programmed with drag & drop features from the software, The PTT button can be programmed for Push To Talk or for latching functionality.

The unit offers “Hold” and “Busy” LED signals in addition to the zone LED’s, and these allow the easy identification of Selection / Busy signals for the user. In addition, to comply with EN 54-16, separate POWER, FAULT and EVAC indicators are provided.

The RS485 communication protocol offers daisy chaining of up to 300 m on a simple CAT5 cable, and

yet makes outlets easy to connect via a standard RJ45 connector.

4.2.8.2 Control Panel



1. Fireman Microphone Hook

   An U-shape hook to place fireman microphone.

2. LED Test Button

   Test LED on control panel, after pressing this button all LEDs light up and then off.

3. Talk Indicator

   This LED light up if PCP is allowed to paging, on the other word, the paging request has been acknowledged.

   ⚠ Sometime after paging request is granted, zones under paging still can be occupied by other sources. In such a case, the talk led still light up, but you can check busy LED to know the situation.

4. Busy Indicator

   Show the status of zone:

Color	Status	Frequency	Activity
Red	Permanent	●	Priority of the zone is lower than other sources.
Green	Blinking	● ◦ ● ◦ ●	Priority of the zone is higher than other sources.
Green	Permanent	●	The zone only desired for paging is not using by other sources.

5. Hold Indicator

   These LEDs Indicate which keys are selected.

6. Monitoring Speaker

This embedded speaker to play chime when paging or monitor signal from audio processor, it needs to work with rs485 output component to get audio signal from other DSP components in audio processor.

7. Fireman Microphone Connector

DIN Connector for fireman microphone connection.

8. All Call Release Button

This button is used to select or deselect all keys.

9. All Call Release Indicator

This indicator lights up if all keys are selected.

10. Event Select Key

These buttons are used to select keys, each key may link to an event for triggering actions like paging or adjust the value of parameters.

11. Power Indicator

This LED light up when the CD8 is powered on.

12. Evacuation Indicator

Light up when system is under evacuation paging.

13. Fault Indicator

To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent		There are faults in audio processor.
Blinking		Lost communication with audio processor.

4.2.8.3 Characteristics

❖ Case

Dimension = 320mm (W) x 190mm (H) x 130mm (D).

Weight = 3.6Kg.

Color = RAL7035.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	120mA	-

❖ Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 1%.

- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖ Front Panel Speaker

- Impedance = 8Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 8kHz.

❖ Cable length

100m(control and power on the same cable)

900m(when power line connects to junction box).

❖ Comes standard with Junction box (JB) and CAT5 cable (1.5)

4.2.9 CD16

4.2.9.1 Overview

CD16 Wall mounted cabinet remote paging console for Ateis audio processor.

Wall-mount heavy duty remote paging console with Access level 2 protection to comply with the EN 54-16. Compatible with Ateis audio processor and communicates over a dedicated RS485 for Power, audio

and DATA.

The unit comprises of 16 zones / 16 buttons with fist firemen's microphone in a metal surface mount wall-box. It provides robust IP-30 protection. Each CD16 contains a PMM PS Master PCB with extension keypad and uses the same architecture as for the PPM AS series of microphone consoles. Each Ateis audio processor like IDA8C can handle up to 31 CD16 units per 485 port in Master/Slave configuration. (Note that only the Master unit is secured, Slaves are not).

The buttons can represent a single zone or a group of zones. All buttons can be programmed with drag & drop features from the software, The PTT button can be programmed for Push To Talk or for latching functionality.

The unit offers "Hold" and "Busy" LED signals in addition to the zone LED's, and these allow the easy identification of Selection / Busy signals for the user. In addition, to comply with EN 54-16, separate POWER, FAULT and EVAC indicators are provided.

The RS485 communication protocol offers daisy chaining of up to 300 m on a simple CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connector.

4.2.9.2 Control Panel

1. Fireman Microphone Hook

   An U-shape hook to place fireman microphone.

2. Busy Indicator(1~8)

   Show the status of zone:

Color	Status	Frequency	Activity
Red	Permanent		Priority of the zone is lower than other sources.
Green	Blinking		Priority of the zone is higher than other sources.
Green	Permanent		The zone only desired for paging is not using by other sources.

3. Fault Indicator

To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent		There are faults in audio processor.
Blinking		Lost communication with audio processor.

4. Monitoring Speaker

This embedded speaker to play chime when paging or monitor signal from audio processor, it needs to work with rs485 output component to get audio signal from other DSP components in audio processor.

5. Fireman Microphone Connector

DIN Connector for fireman microphone connection.

6. Talk Indicator

This LED light up if PCP is allowed to paging, on the other word, the paging request has been acknowledged.

Sometime after paging request is granted, zones under paging still can be occupied by other sources. In such a case, the talk led still light up, but you can check busy LED to know the situation.

7. LED Test Button

Test LED on control panel, after pressing this button all LEDs light up and then off.

8. Hold Indicator(1~8)

These LEDs Indicate which keys are selected.

9. All Call Release Button

This button is used to select or deselect all keys.

10. All Call Release Indicator

This indicator lights up if all keys are selected.

11. Event Select Key(1~8)

These buttons are used to select keys, each key may link to an event for triggering actions like paging or adjust the value of parameters.

12. Busy Indicator(9~16)

Please refer to Busy Indicator(1~8).

13. Hold Indicator(9~16)

Please refer to Hold Indicator(1~8).

14.Event Select Key(9~16)

Please refer to Event Select Key(1~8).

15.Power Indicator

This LED light up when the CD8 is powered on.

16.EVAC Indicator

Light up when the system is under evacuation paging.

4.2.9.3 Characteristics

❖Case

Dimension = 350mm (W) x 250mm (H) x 130mm (D).

Weight = 4.7Kg.

Color = RAL7016

❖Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	150mA	

❖Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 0.1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖Front Panel Speaker

- Impedance = 8Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖Cable length

100m(control and power on the same cable)

900m(when power line connects to junction box).

❖Comes standard with Junction box (JB) and CAT5 cable (1.5)

4.2.10 Fireman Microphone

4.2.10.1 Overview

The Fireman Microphone SHM 1 is a small compact microphone dedicated to security calling. It is a totally monitored microphone with internal resistor network to detect four different states (Shortcut, Closed Contact, Open Contact, Absent). Firemen Microphone SHM 1 for All call, group or zone calls.

4.2.10.2 Characteristics

❖Case

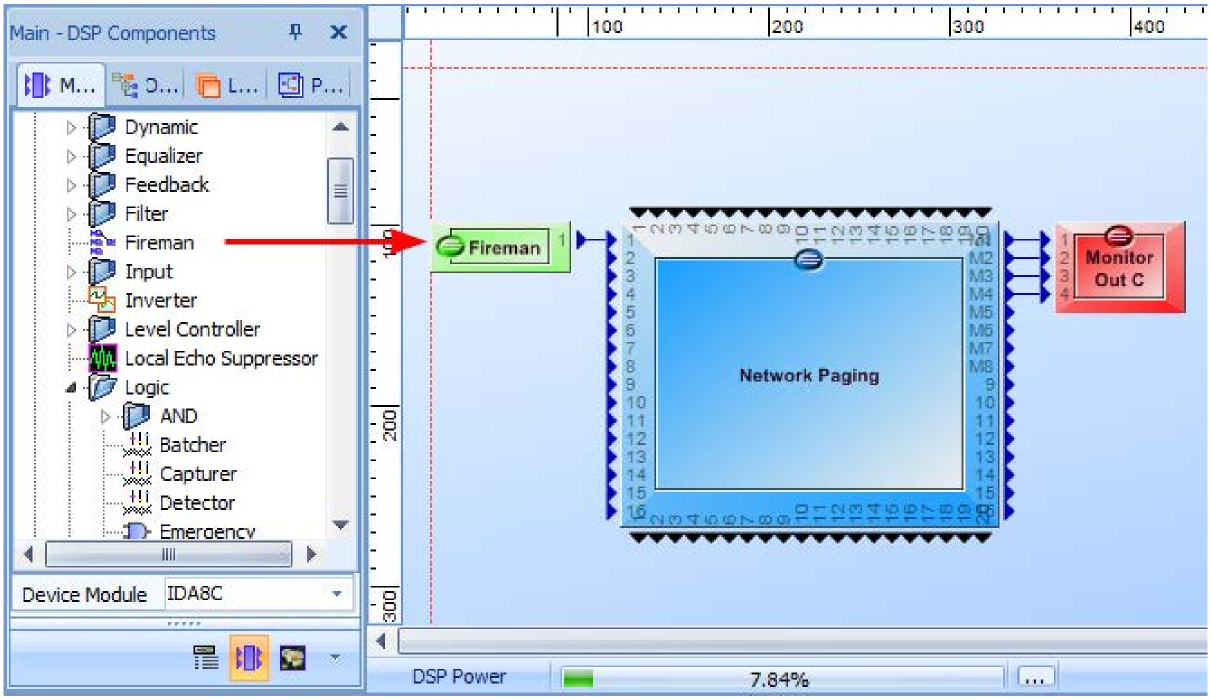
- Dimension = 62mm (W) x 45mm (L) x 102mm (H).
- Weight = 0.15KG
- Cable = 1.8M(spring cord)

❖Microphone

- Bandwidth @ -3dB = 300Hz ~ 6kHz.
- Impedance = 500ohm.
- Sensitivity = -72dB

4.2.10.3 Configuration

Fireman microphone needs config using Ateis Studio before working. Fireman is the DSP component for the microphone. Most of time Fireman component is working with Network Paging component, in such settings, fireman microphone is able to paging.



See more details in topic Fireman component.

4.2.11 CDPA

4.2.11.1 Overview

CDPA Wall mounted cabinet remote paging console for Ateis audio processor.

Wall-mount heavy duty remote paging console with Access level 2 protection to comply with the EN 54-16. Compatible with Ateis audio processor and communicates over a dedicated RS485 for Power, audio and DATA.

CDPA supports 2 channels of music input. Users can press the button for music channel selection. There are two LED indicators to show which music channel is active.

The unit comprises of 24 zones / 24 buttons with fist firemen’s microphone in a metal surface mount wall-box. It provides robust IP-30 protection. Each CDPA contains two PMM PS Master PCBs with extension keypads and uses the same architecture as for the PPM AS series of microphone consoles. Each Ateis audio processor like IDA8C can handle up to 31 CDPA units per 485 port in Master/Slave configuration. (Note that only the Master unit is secured, Slaves are not).

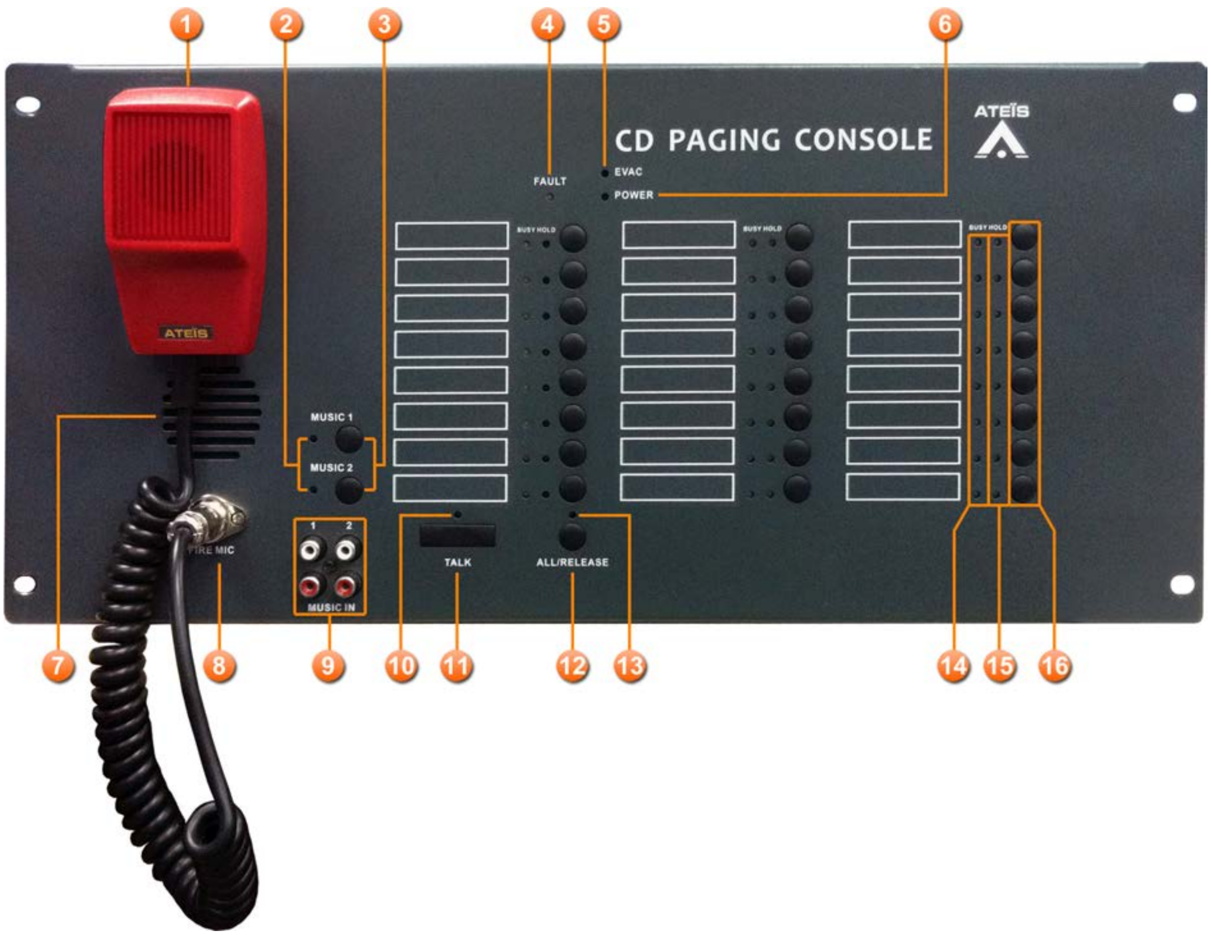
The buttons can represent a single zone or a group of zones. All buttons can be programmed with drag & drop features from the software, The PTT button can be programmed for Push To Talk or for latching functionality.

The unit offers “Hold” and “Busy” LED signals in addition to the zone LED’s, and these allow the easy identification of Selection / Busy signals for the user. In addition, to comply with EN 54-16, separate POWER, FAULT and EVAC indicators are provided.

Additional RCA connectors with selection buttons support local audio injection for commercial usage.

The RS485 communication protocol offers daisy chaining of up to 300 m on a simple CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connector.

4.2.11.2 Control Panel



1. Fireman Microphone

   The Fireman Microphone SHM 1 is a small compact microphone dedicated to security calling. It is a totally monitored microphone with internal resistor network to detect four different states (Shortcut, Closed Contact, Open Contact, Absent). Firemen Microphone SHM 1 for All call, group or zone calls.

2. Music Active Indicator

   These two led to indicate which music input channel is active.

3. Music Selection Button

   To select music input channel.

4. Fault Indicator

   To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent		There are faults in audio processor.
Blinking		Lost communication with audio processor.

5. EVAC Indicator

   Light up when the system is under evacuation paging.

6. Power Indicator

   This LED light up when the CD8 is powered on.

7. Monitoring Speaker

   This embedded speaker to play chime when paging or monitor signal from audio processor, it needs to work with rs485 output component to get audio signal from other DSP components in audio processor.

8. Fireman Microphone Connector

   DIN Connector for fireman microphone connection.

9. Music Input Connector

   There are two music input channels of CDPA. Users can select one of them by pressing music selection button.

10. Talk Indicator

   This LED light up if PCP is allowed to paging, on the other word, the paging request has been acknowledged.

   Sometime after paging request is granted, zones under paging still can be occupied by other sources. In such a case, the talk led still light up, but you can check busy LED to know the situation.

11. Talk Button

   Press this button to request paging. There are two modes for this button: push to talk and lock.

12. All Call/Release Button

   This button is used to select or deselect all keys.

13. All Call/Release Indicator

   This indicator lights up if all keys are selected.

14. Busy Indicator(1~8)

   Show the status of zone:

Color	Status	Frequency	Activity
Red	Permanent		Priority of the zone is lower than other sources.
Green	Blinking		Priority of the zone is higher than other sources.
Green	Permanent		The zone only desired for paging is not using by other sources.

15. Hold Indicator(1~8)

These LEDs Indicate which keys are selected.

16.Event Select Key(1~8)

These buttons are used to select keys, each key may link to an event for triggering actions like paging or adjust the value of parameters.

4.2.11.3 Characteristics

❖Case

Dimension = 483 mm (W) x 220 mm (H) x 68 mm (D).

Color = RAL7035.

❖Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	200mA	

❖Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 0.1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖Front Panel Speaker

- Impedance = 8Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖MUSIC INPUT

- Input Impedance = 10kOhm.
- Maximum level = 12dBu
- THD @ 1kHz < 0.1%.
- Bandwidth @ -3dB = 20Hz ~ 20kHz.

❖Cable length

100m(control and power on the same cable)

900m(when power line connects to junction box).

❖Comes standard with Junction box (JB) and CAT5 cable (1.5)

4.2.12 PCP

4.2.12.1 Overview

Wall-mount heavy duty remote paging console with Access level 2 protection to comply with the EN 54-16. Compatible with Ateis audio processor and communicates over a dedicated RS485 for Power, audio and DATA.

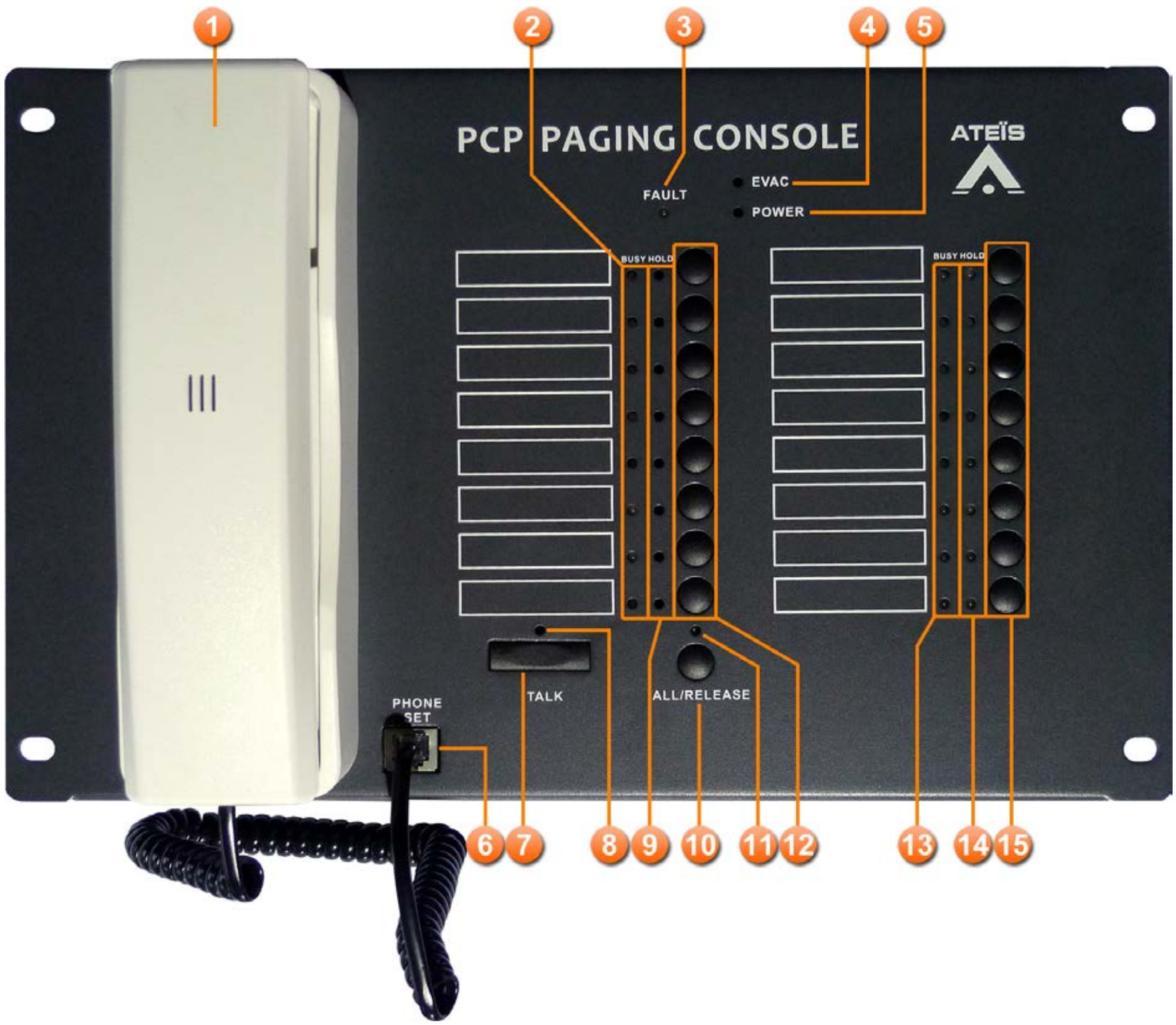
The unit comprises of 16 zones / 16 buttons with fist firemen's microphone in a metal surface mount wall-box. It provides robust IP-30 protection. Each PCP contains a PMM PS Master PCB with extension keypad and uses the same architecture as for the PPM AS series of microphone consoles. Each Ateis audio processor like IDA8C can handle up to 31 PCP units per 485 port in Master/Slave configuration. (Note that only the Master unit is secured, Slaves are not).

The buttons can represent a single zone or a group of zones. All buttons can be programmed with drag & drop features from the software, The PTT button can be programmed for Push To Talk or for latching functionality.

The unit offers "Hold" and "Busy" LED signals in addition to the zone LED's, and these allow the easy identification of Selection / Busy signals for the user. In addition, to comply with EN 54-16, separate POWER, FAULT and EVAC indicators are provided.

The RS485 communication protocol offers daisy chaining of up to 300 m on a simple CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connector.

4.2.12.2 Control Panel



1. A telephone styled microphone for paging.

2. Busy Indicator(1~8)

Show the status of zone:

Color	Status	Frequency	Activity
Red	Permanent	●	Priority of the zone is lower than other sources.
Green	Blinking	● ° ● ° ●	Priority of the zone is higher than other sources.
Green	Permanent	●	The zone only desired for paging is not using by other sources.

3. Fault Indicator

To display fault status of system. Following table list the states of fault:

Status	Frequency	Activity
Permanent	●	There are faults in audio processor.
Blinking	● ◦ ● ◦ ●	Lost communication with audio processor.

4. EVAC Indicator

   Light up when the system is under evacuation paging.

5. Power Indicator

   This LED light up when the PCP is powered on.

6. Microphone Connector

   A phone jack connector for connection of telephone styled microphone.

7. Talk Button

   Press this button to request paging. There are two modes for this button: push to talk and lock.

8. Talk Indicator

   This LED light up if PCP is allowed to paging, on the other word, the paging request has been acknowledged.

   ⚠ Sometime after paging request is granted, zones under paging still can be occupied by other sources. In such a case, the talk led still light up, but you can check busy LED to know the situation.

9. Hold Indicator(1~8)

   These LEDs Indicate which keys are selected.

10. All Call/Release Button

    This button is used to select or deselect all keys.

11. All Call/Release Indicator

    This indicator lights up if all keys are selected.

12. Event Select Key(1~8)

    These buttons are used to select keys, each key may link to an event for triggering actions like paging or adjust the value of parameters.

13. Busy Indicator(9~16)

    Please refer to Busy Indicator(1~8).

14. Hold Indicator(9~16)

    Please refer to Hold Indicator(1~8).

15. Event Select Key(9~16)

    Please refer to Event Select Key(9~16).

4.2.12.3 Characteristics

❖ Case

Dimension = 346 mm (W) x 220 mm (H) x 128 mm (D).

Weight = 3 Kg.

Color = RAL7035.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	150mA	

❖ Handset Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 0.1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖ Front Panel Speaker

- Impedance = 8Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖ Cable length

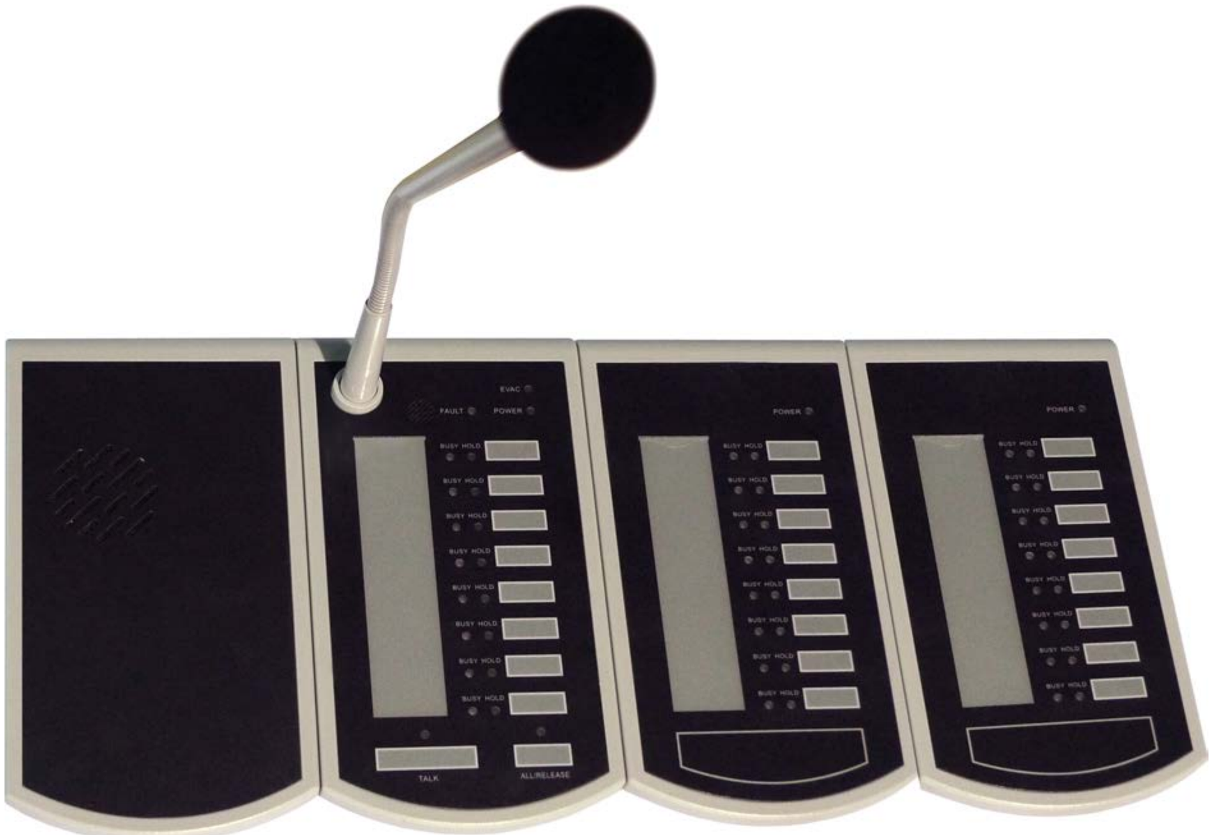
100m(control and power on the same cable)

900m(when power line connects to junction box).

❖ Comes standard with Junction box (JB) and CAT5 cable (1.5)

4.2.13 PSC

4.2.13.1 Overview



Unidirectional Condenser Addressable Microphone, compatible with all system units, PSC uses an RS485 protocol over a single CAT5 cable connection, to transport both Audio and Power from the PSC to the system units. The PSC comprises of 8 zones / 8 buttons with sleek condenser gooseneck microphone, and spring metal protection, providing durability and excellent aesthetics as well as allowing up to 256 zones expansion via the additional keypad easy extension station. The buttons can represent a single zone or a group of zones and are easily defined via the GUI of the system units using a simple Matrix selection.

It supports an extended speaker to monitor the audio source with more power compared with the original one.

The unit offers “Hold” and “Busy” LED signals in addition to the zone LED’s, and these allow the easy identification of selection / Busy signals for the user.

All buttons can be programmed with drag & drop features from the System unit GUI software and each button can be programmed for Push To Talk or Latch functionality.The unidirectional condenser microphone warrants high quality directive signal pick up from the user and hence less interference from the surroundings thanks to the cardioid polar pick-up pattern.

The RS485 communication protocol offers daisy chaining of up to 100 m on a CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connectors. (The microphone compatibility listing shows the maximum number of units per System).

4.2.13.2 Characteristics

❖Case

Dimension = 220mm (W) x 483mm (L) x 115.7mm (H).

Weight = 0.37 Kg.

Color = RAL7035.

❖Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	150mA	

❖Microphone Output

- Maximum level = 6dBu.
- Output Impedance = 100Ohm.
- THD @ 1kHz < 0.1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖Front Panel Speaker

- Impedance = 8Ohm.
- Maximum Power @ 1kHz = 1W.
- THD @ 1kHz < 1%.
- Bandwidth @ -3dB = 200Hz ~ 20kHz.

❖Cable length

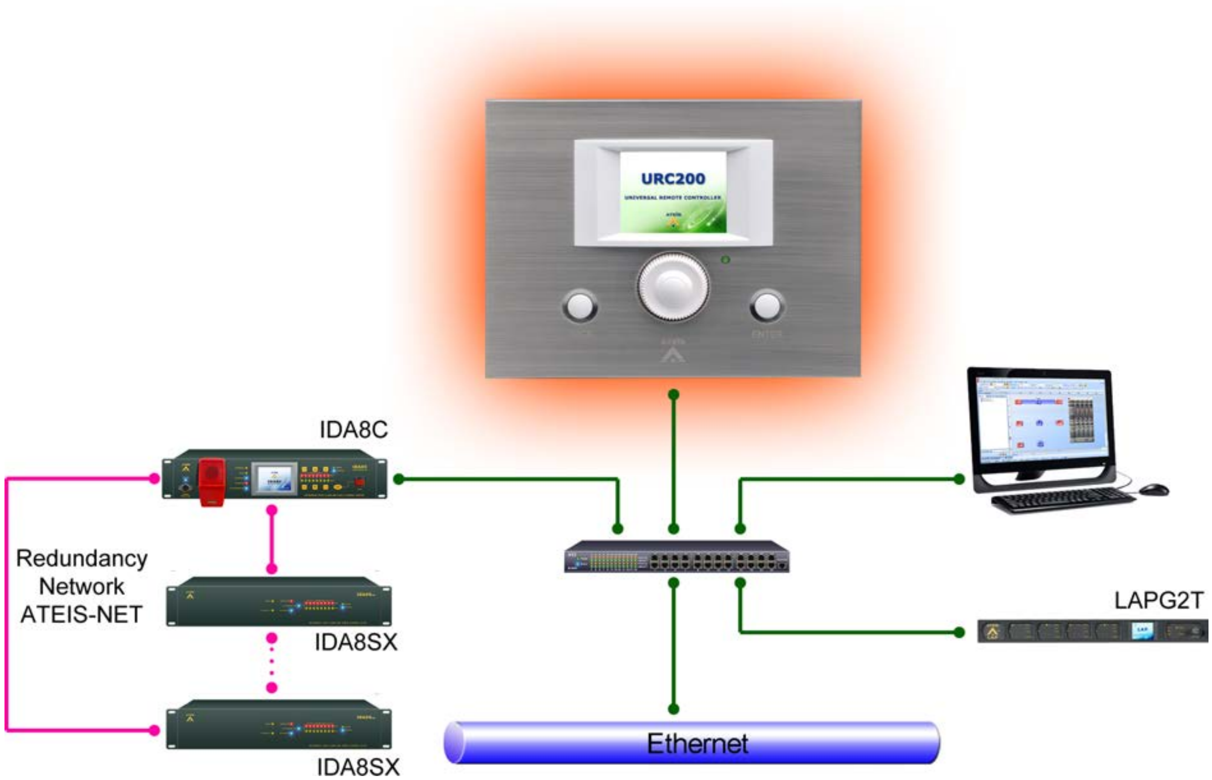
100m(control and power on the same cable)

900m(when power line connects to junction box).

❖Comes standard with Junction box (JB) and CAT5 cable (1.5)

4.2.14 URC200 TPC

4.2.14.1 Overview



The URC200 is an programmable remote controller (TCP/IP) for the IDA8 system, LAPG2T and ECS DSP audio matrix System. It allows users to customize the menu items for the control parameters, master/sub presets of Ateis audio processors.

The URC200 contains a color 2" TFT panel, and with the well designed user interface it gives the user the best experience of using this control console. Two buttons and one encoders make menu controlling very simple and easy.

The URC200 is powered over IP and easy to integrate with current demands for room controllers like light, curtains, sound and video control. The full color display is easy to read and has a low-power consumption to allow for long lines and multiple devices into one system.

4.2.14.2 Control Panel

1. TFT Panel

   A color 2" TFT panel presents the menu system of URC 200 TPC.

2. IR Receiver

   It receives the control signals of handheld devices for controlling the system.

3. Back Button

   This button is used for going back to the upper layer of the menu.

4. Encoder

   Rotate this encoder for select the item of the menu.

5. Enter Button

   This button is used for going into to the inner layer of the menu.

4.2.14.3 Characteristic

❖ Case

Dimension = 140mm (W) x 108mm (L) x 34mm (H).

Weight = 0.35Kg.

Color = RAL7016.

❖ Screen

Diagonal = 2".

Resolution = 176 x 220.

❖ Power

Item	Voltage	Current Consumption	Comment
DC Input	18V~26V, Typical 24V	50mA	

❖Maximum Cable Length

100m on Category 5 cable

4.2.14.4 Configuration

This topic with sub-topics which describe configuration of URC200 TPC. This is the basic procedure to setup URC200 TPC for controlling the system:

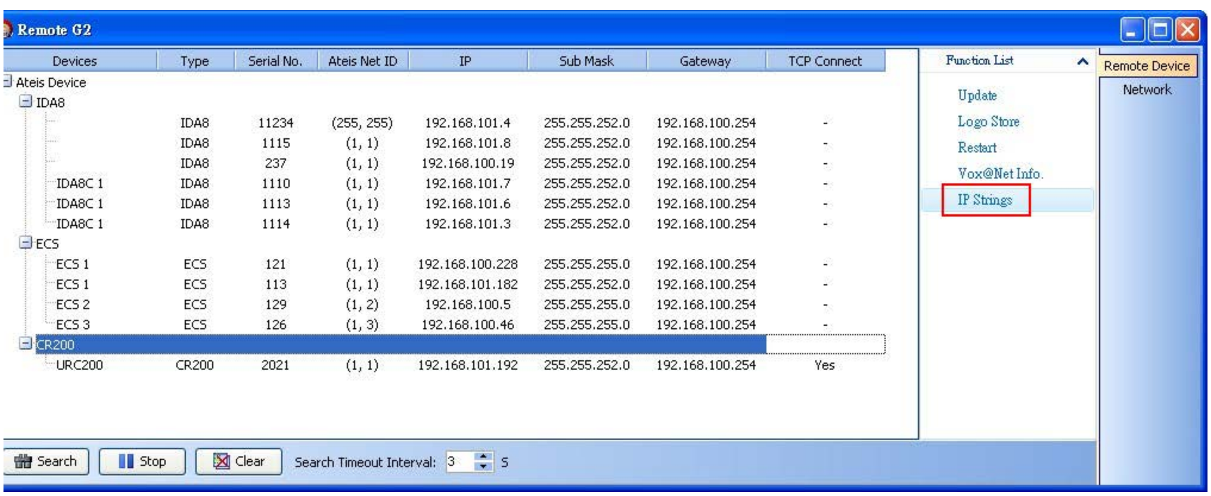
1. Connect URC200 TPC to ethernet switch that the targeted devices connected. The target devices are IDA8C, ECS, ...etc.
2. Set IP/Subnet Mask/Gateway in Protected Mode menu.
3. Edit Control Items, see topic Edit Control Items for more detail.
4. Save Configuration to URC200 TPC, see topic Save/Load Configuration for more detail.

4.2.14.4.1 Edit Control Items

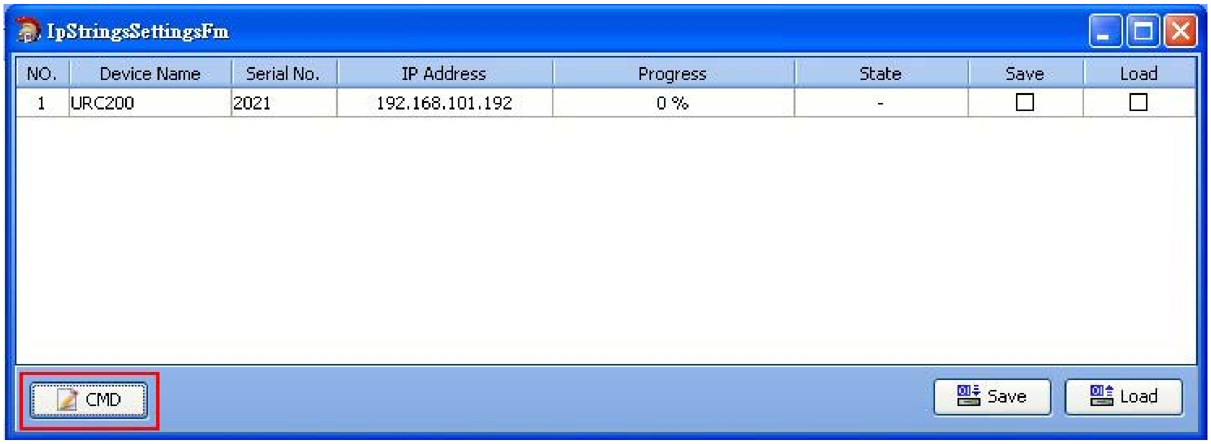
The menu system of URC200TPC is configurable, that means users need to create items of menu for what they want to control. The menu system is multi-layered, it the first layer, there are three items: ECS, LAP G2, AMP indicates what device can be controlled by URC200 TPC.

Following steps is an instruction for building menus.

1. Lunch the Remote G2 software. On the left, there is a grid lists all the devices could be found by it, select the node "CR200" and click [IP settings] in [Remote Device] page.

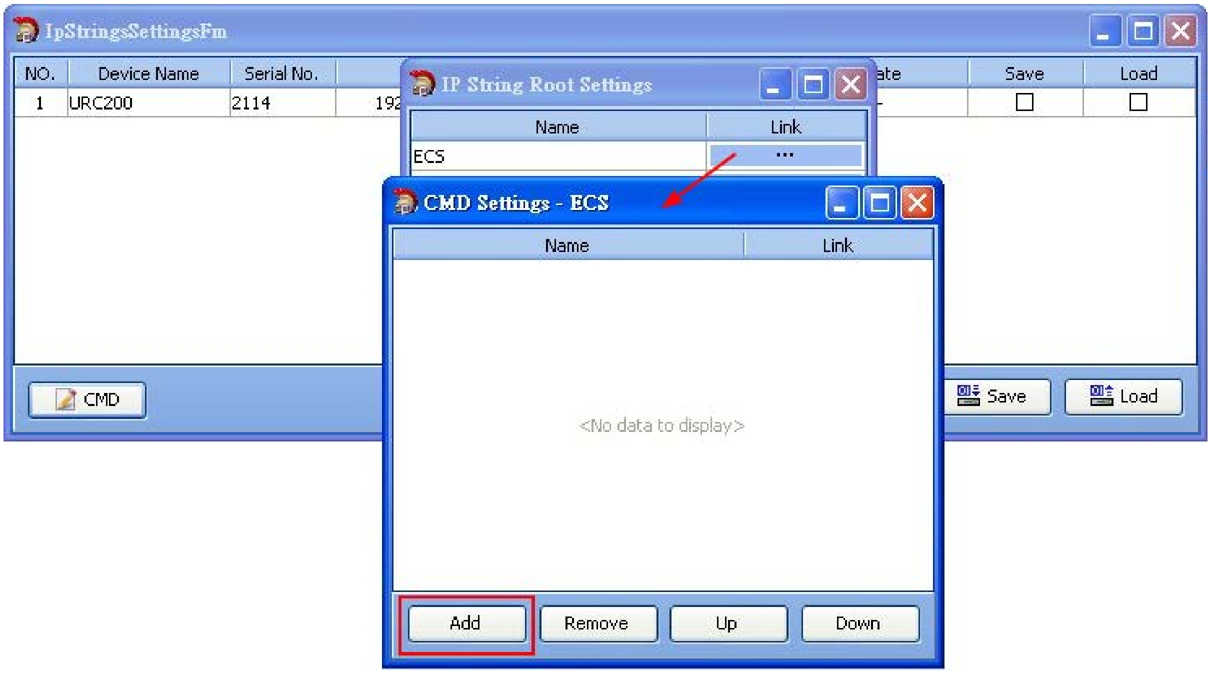


2. It open the window [IpStringsSettingsFm]. This window lists all the URC200 TPC that ready to save or load the configuration. Click the button [CMD] to open the settings window for the first layer items in the menu system.

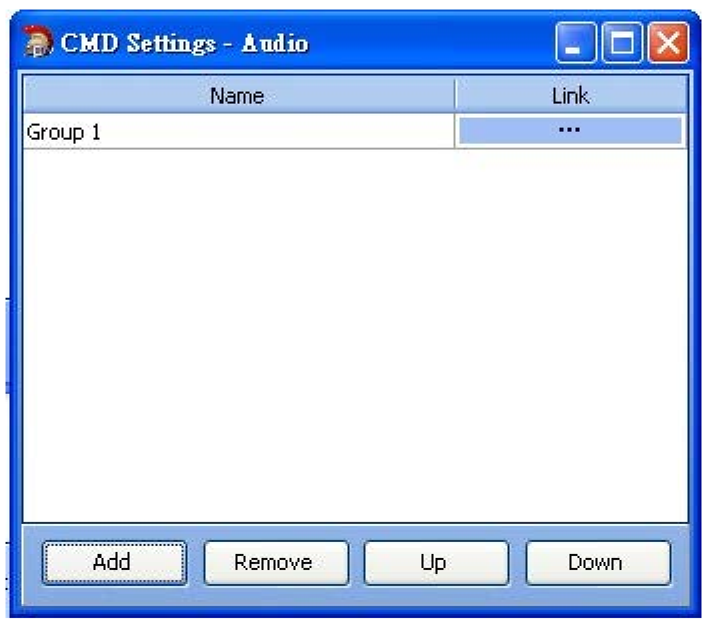


3. Press button [Reset] to make the reset root menu to default.

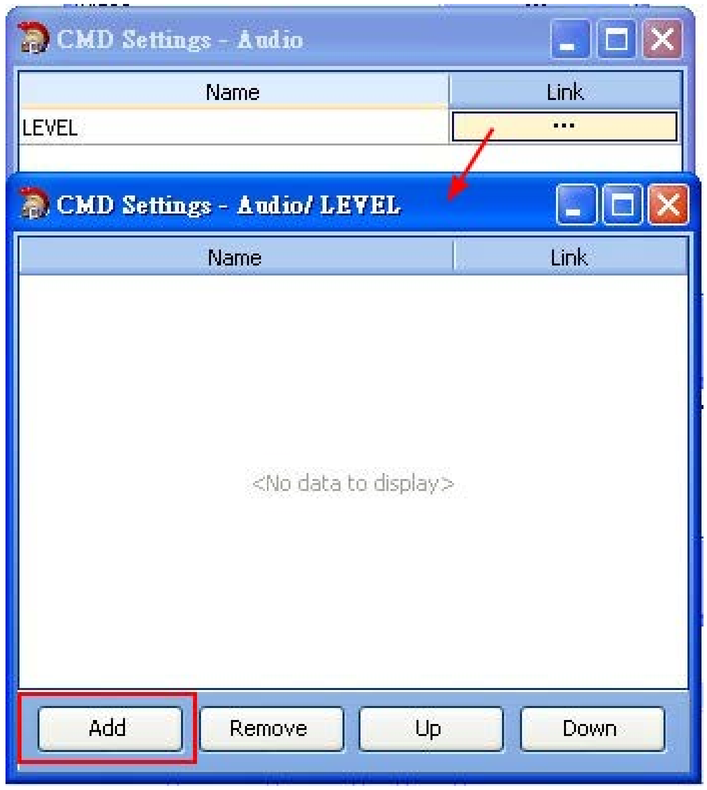
4. Click the button on field [Link] for the desired menu item. It opens the CMD settings window. Press button [Add] to establish an item in the menu for controlling parameter of the device.



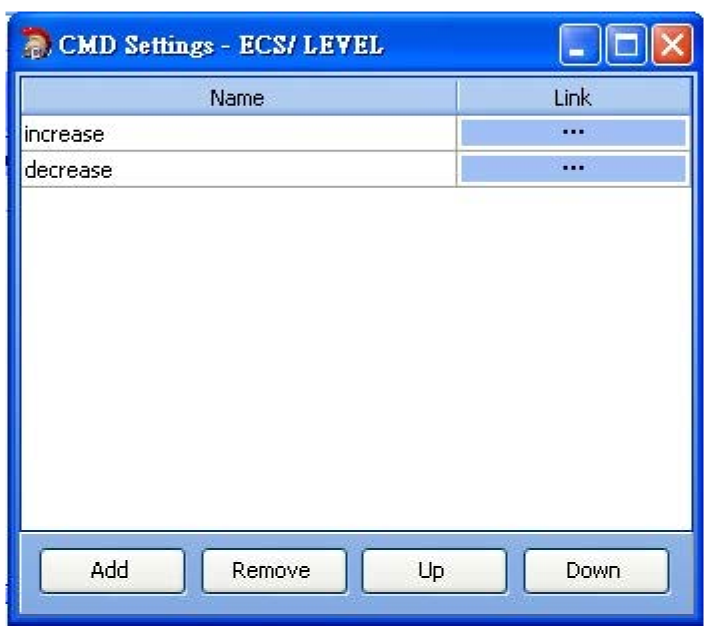
5. The default name is "Group 1", you can change it to the other string.



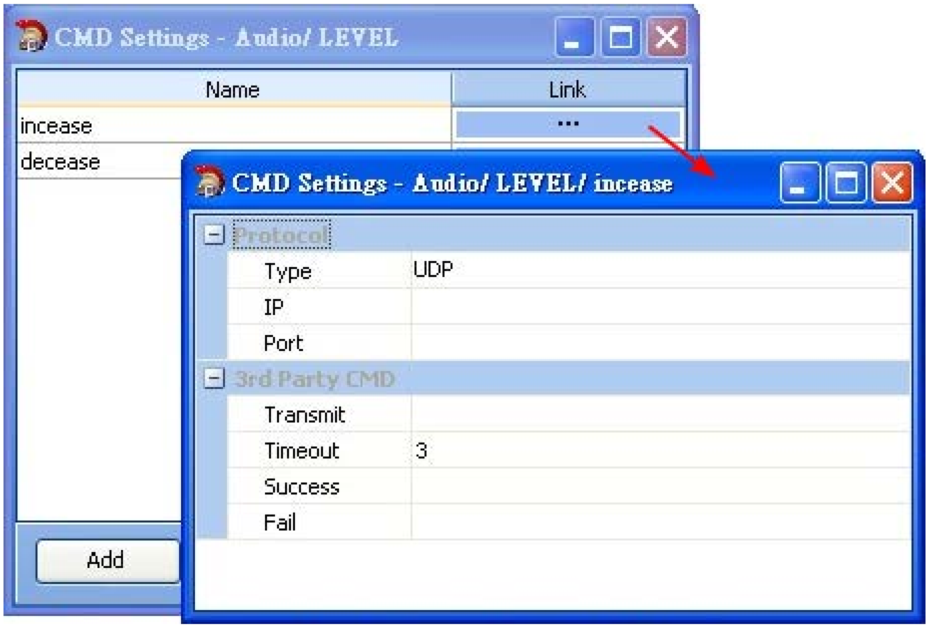
6. Click buttons on the field [Link] to open the command list window. Press button [Add] to create a command string.



7. For example, I create two items: "increase" and "decrease" for controlling the audio level.



8. Clink [link] field to fill the detail settings of the command string.



The fields in the grid are described below:

- Type

  To indicate the communication protocol used, this field is read only.

- IP

  The IP address of the device be controlled.

- Port

  The port number of the device be controlled.

- Transmit

  The command string for controlling the parameter of the target device. When the user selects the item of menu and press button [Enter] on the panel, this string will send to the target device.

- Timeout

  Due to the networking quality could be varied in the different environment. For some applications with low bandwidth or bad quality network, we can set the timeout value to a large number because it may take more time to communicate with target devices. 3 seconds is the default value and it is suitable for most cases.

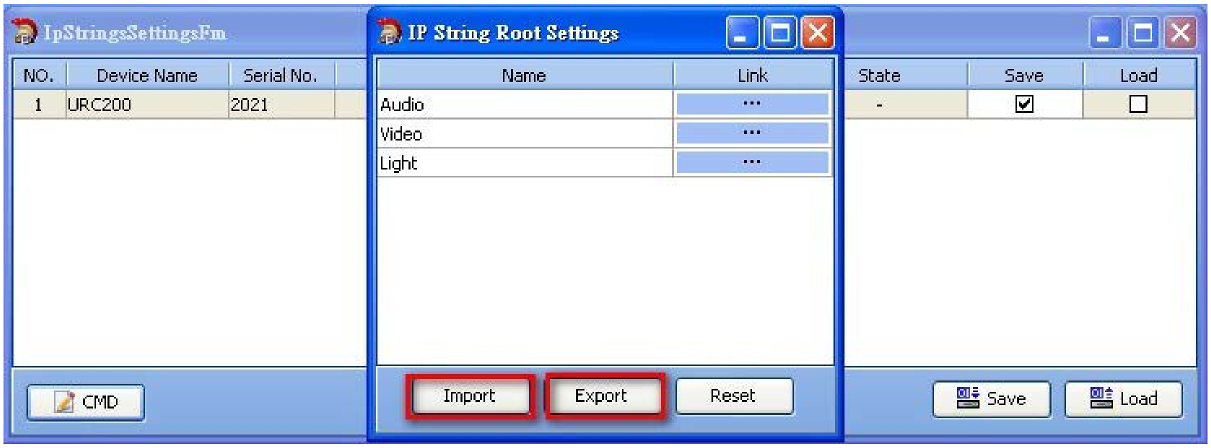
- Success

  Set this field could show "Success" when the target device informs URC200 TPC the command string is transferred successfully. This setting is optional, if you need it, please fill the value "06 0D".

- Fail

  Set this field could show "Fail" when the target device informs URC200 TPC the command string is transferred failed. This setting is optional, if you need it, please fill the value "10 0D".

4.2.14.4.2 Import/Export Configuration



You can export the URC200 TPC configuration to a file or import the file to restore the configuration. On settings window for the first level menu, there is two buttons:

- Export

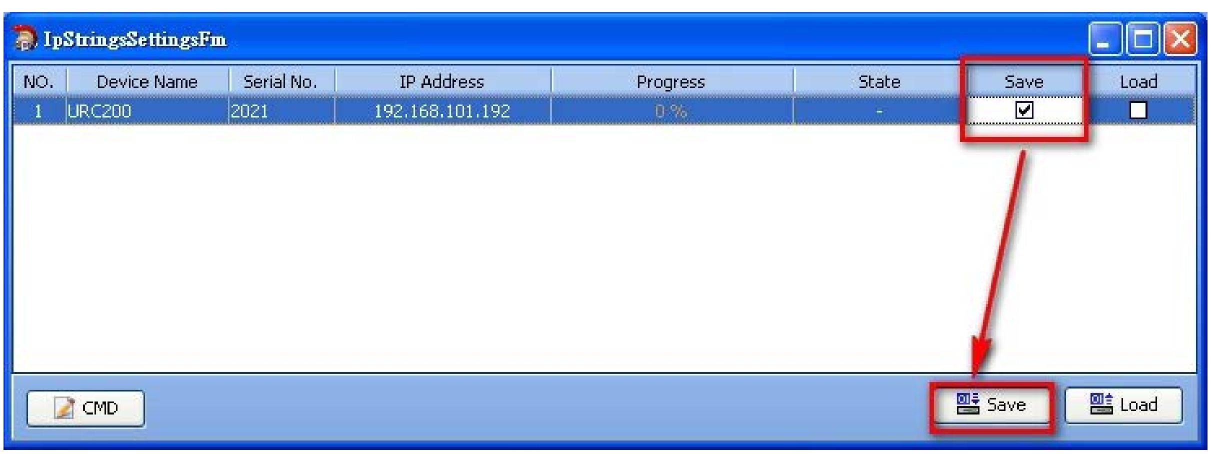
  By clicking this button to export configuration of URC200 TPC to a file.

- Import

  By clicking this button to import configuration of URC200 TPC from a file.

4.2.14.4.3 Save/Load Configuration

The configuration of URC200 TPC is edited using the software. If you want URC200 TPC to run the configuration, you need to save it to the device. See the below figure, you have to select the target URC200 TPC that to be saved and then press the button [Save] to start save procedure. During the save period, the field [progress] in the grid shows the percentage the save procedure done.



4.2.14.4.4 Reset Configuation

Sometimes if URC200 TPC power on with error, it needs to ignore current configuration, and reset to the default status. Holds on both two keys on the panel, and plugs the ethernet wire to URC200 TPC to power on. See the following demonstration.

After power up, you'll the a message on the top of LCD screen: "No Preset". It means the URC200 TPC now is forbidden to run the configuration. And then you can save another configuration for the device to make it working.

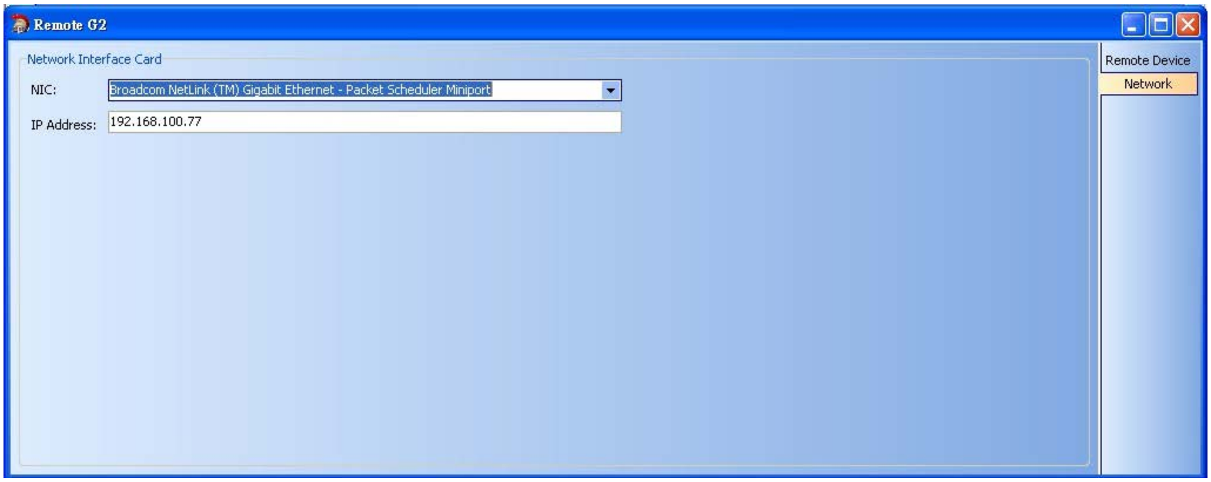
4.2.14.5 Device Maintenance

This topic contains the knowledge about the update procedure and inner settings of devices. For example the IP settings.

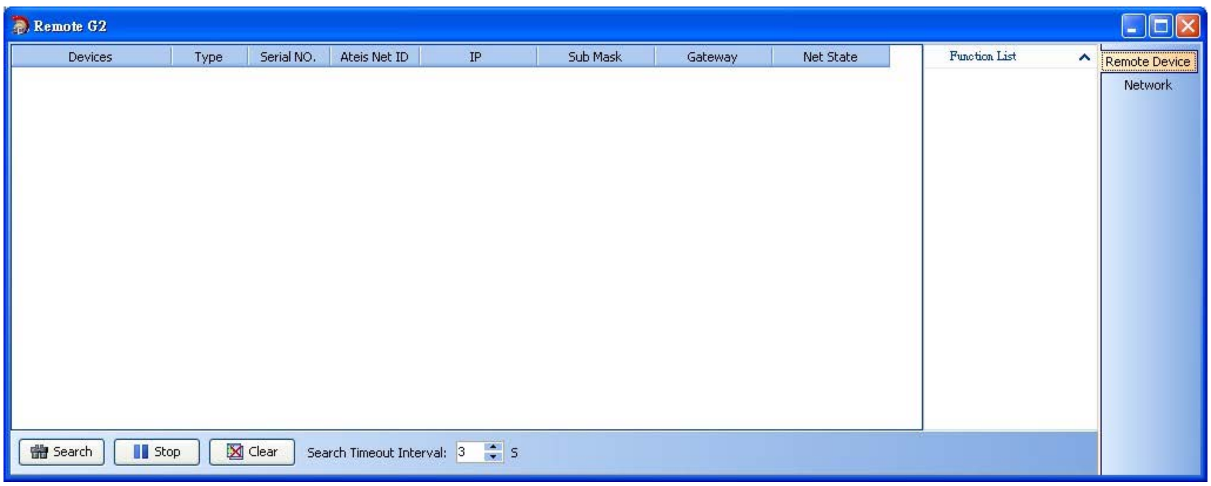
4.2.14.5.1 Update Firmware

Please follow the steps below to update the firmware of URC200TPC:

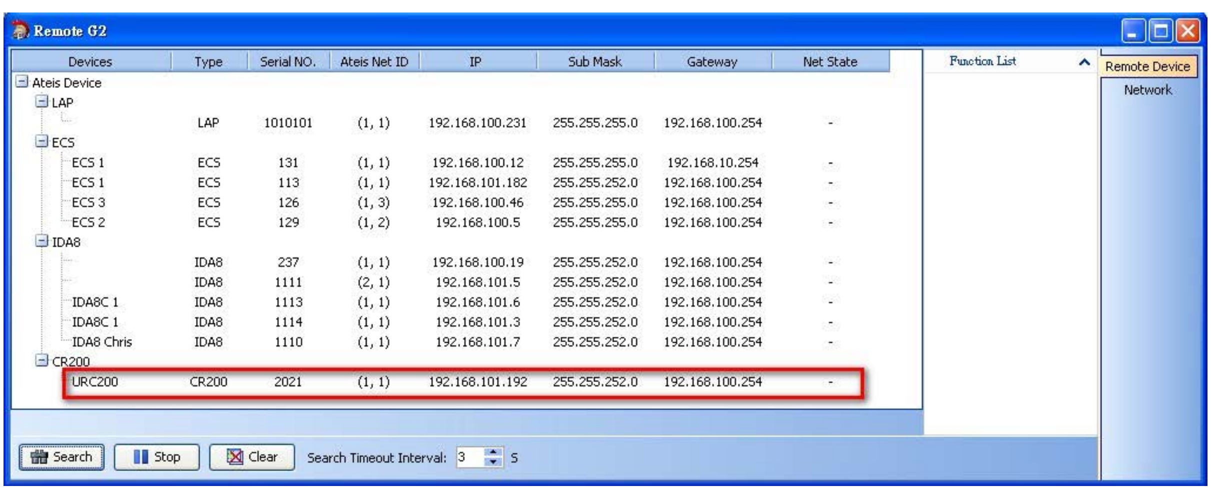
1. Lunch software and search URC200TPC. Click page [Network]. Set the network interface card that communicate with URC200TPC. After the network interface card is selected, the IP address is refreshed automatically.



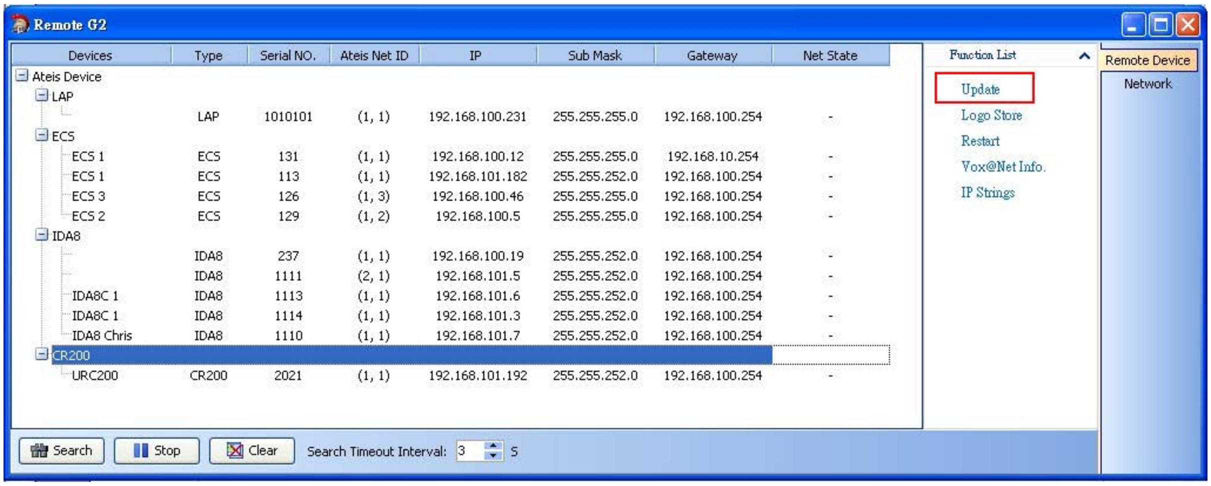
2. Click page [Remote Device]. This page lists all devices in the network. Press the button [Search] to discovery URC200 TPCs throughout the network.



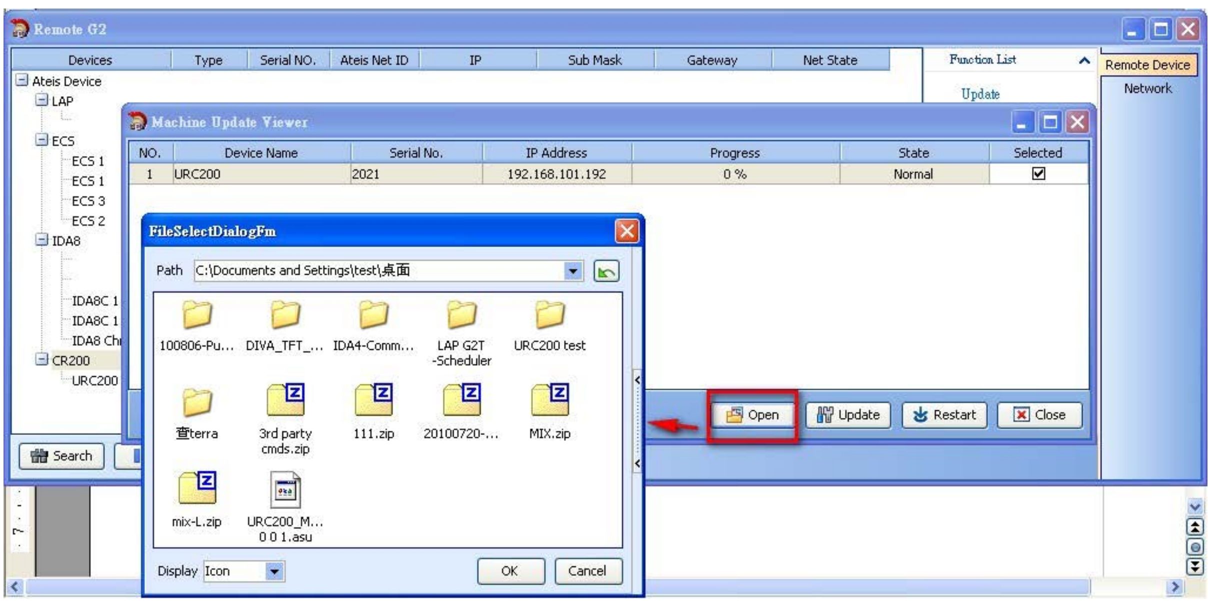
3. The result of a search is like below. In the field [Devices], devices are represented using a tree, each leaf node of the tree is a device. The URC200 is one of them in the grid.



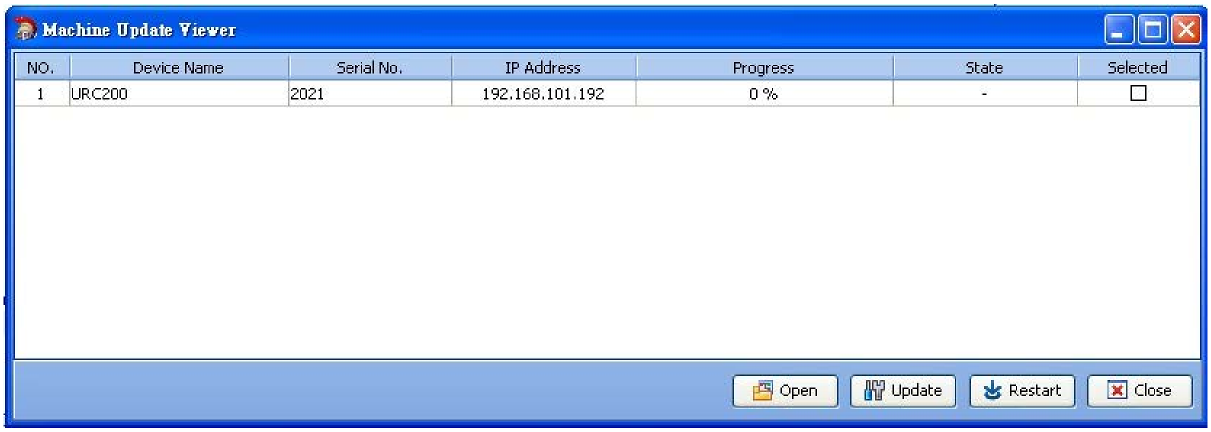
4. There is a box [Function List] at the right side of the window. By select the entry of the grid, the items inside the box changes accordingly to list all the functions that the device provide. Select entry [CR200], and then click the item [Update] in the [Function List] box to open the update window.



5. Click the button [Open], use the dialog window to choose the desired firmware program for updating.



6. Select the desired URC200 TPC for updating by clicking the check box in the field [Selected]. Press the button [Update] to start updating procedure, which uploads the firmware program to the device's flash memory. There are progress bars for each URC200 TPC displaying the progress of the procedure. Once updating is finished, please press the button [Restart] to restart the device and make the change takes effect.



4.2.14.5.2 Protected Mode

There are some internal settings of URC200 TPC. Those settings are in the protected mode menu. To enter the protected mode, please hold both two buttons and turn the knob.

The internal settings are:

❖ Network

This page contains following settings:

- IP Address
- Sub Mask
- Gateway

Basically, those values should be defined before using a URC200 TPC.

❖ Version

This page displays the firmware version of URC200 TPC.

❖ S. B.

This page is comprised of two settings:

- Saver Time:

  The waiting time that the menu go back to root page.

- Black Time:

  The waiting time that the URC200 TPC enters sleep mode.

Note: The unit of the value is minute.

❖ Reset

If you modify the settings belong to protected mode, please execute this function to save value and take effect.

❖ Return

Execute this function to go back root menu.

4.2.15 PPM-IT5

4.2.15.1 Overview

The PPMIT5 IP paging console is a man-machine interface which allows call-paging, messages broadcasting. Its back-lit touch screen is designed for simple and user-friendly operating. The 3 hardware keys can be freely assigned by software. The PPM-IT5 Media console is a versatile device that fits well in a commercial shopping center as for an industrial environment where paging over IP-networking brings flexibility and easy access.

All paging parameters needed for site operating can be programmed: zones assigned to the different buttons, name of zones, group of zones, messages triggering or event control. A total of 168 keys over 12 pages allow zone or group of zones selections. The prerecorded messages and the chime are stored into the PPM-IT5 IP paging console.

All the settings are done through web pages with your favorite web browser. Thanks to powerful echo cancellation, the PPM-IT5 delivers clear sound for full duplex talk.

❖PPM-IT5 is made of:

- 5” TFT full color paging console.
- High quality gooseneck microphone.
- Built in loudspeaker.
- Ethernet interface including PoE (Power Over Ethernet).
- 24 VDC power supply (if no PoE available).
- Automatic gain control on microphone input
- Echo cancellation / noise reduction.
- G.711, G.722, MP3 audio encoding /decoding.
- Half or full duplex talk.
- Memory space for prerecorded messages.
- POWER / FAULT / EVAC LEDs.
- 3 key-buttons: User definable using ATEÏS Studio GUI.
- 168 Touch fields: 14 pages with 12 keys.
- RJ 9 for optional telephone headset.
- 2 mini-jack plugs for optional headset.

❖Characteristics of PPM-IT5 is same as PSS AS, the only difference between them is the power supply, PPM-IT5 can use PoE.

❖Configuration: To make PPM-IT5 work, it needs to edit configuration using Ateis Studio, you can't find this item in Remotes. The right way is using VoIP component.

4.3 Amplifier

4.3.1 SPA

4.3.1.1 Overview

The SPA (Security Power Aamplifier) is designed for perfect integration into the SINAPS system, but thanks to its flexibility, it can also be used for any Public Address application.

SPA 2060

SPA 2120

SPA 2240

The SPA amplifiers were specifically developed to meet the requirements of EN 60849 for safety installations. Each amplifier module is fitted with its own 220 VAC/24 VDC power supply for increased system reliability. To avoid handling errors, the volume output is set using the potentiometer located at the back of the apparatus. In addition to standard protection via fuses, SPA amplifiers also have electronic and thermal protection to protect them from all potential hazards.

A fan provides forced cooling for the final power shelves and internal parts of the apparatus. It starts up automatically when the temperature reaches a certain threshold, and stops when it returns to normal values.

Four LEDs display the status of each amplifier: AC or DC power supply, Line Overload and Temperature overload.

Furthermore, a three LEDs Vu-Meter indicates the presence and level of audio signal.

A surveillance circuit continuously monitors the temperature and the presence of AC and DC power supplies for both amplifiers. In case of a problem, the overload LEDs will be switched ON and the Fault relay will be activated.

SPA series(SPA 2060, 2120, 2240) can be ordered with optional 115V power supply. Please contact your local dealer or your regional ATEÏS branch office.

❖Specifications

- Output power:
  - SPA 2060 : 2 X 60W
  - SPA 2120 : 2 X 120W
  - SPA 2240 : 2 X 240W
- Output voltage: 8 Ù, 70 & 100 V
- Power indicate 24 V and 230 V: Yes
- Frequency respond: 60 -18000 Hz
- Distortion: < 1%
- Output: 4-8 Ù 70-100 V
- Fault contact(s): yes
- Signal indication: Yes

4.3.1.2 Front Panel

❖Front Panel

A VU Meter

The VU-meter is composed of three LEDs that represent three different level thresholds: -40dB, -20dB and 0dB. When the input signal reaches those thresholds, the corresponding LED will blink.

B Overload

The overload LED will blink as soon as the output power exceeds the nominal value. If the Overload LED is blinking, check that the output load is well connected and still in good condition as well as

checking the level of the input signal.

C Temp.

If the device internal temperature exceeds 95 degrees C, the TEMP LED will light and the pre-amp will stop working until the temperature falls below 95 degrees C. The pre-amp will come back to a working state automatically. To avoid temperature failure, please insert an external fan to the rack system.

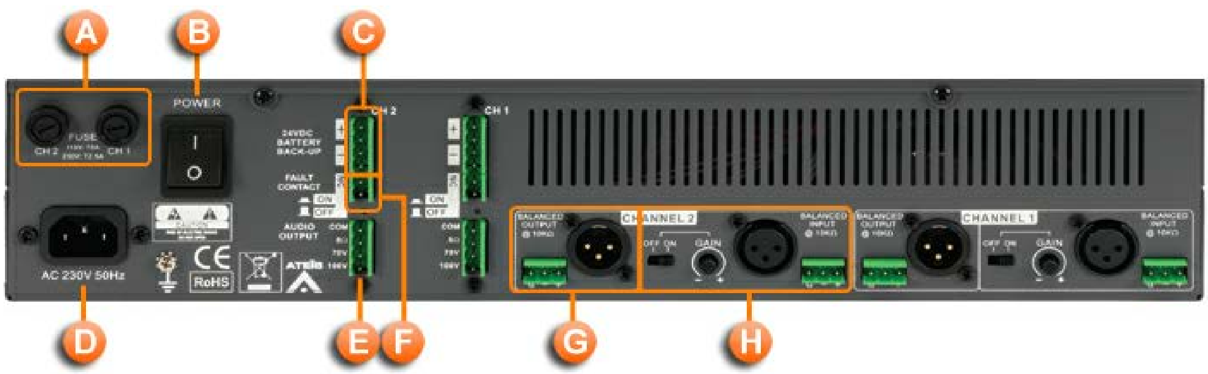
D DC

The DC LED indicates that the SPA amplifier is powered by a 24VDC power supply.

E Power

The Power LED indicates the ON/OFF status of the SPA amplifier.

4.3.1.3 Rear Panel



A Fuse Receptacles

Two fuse receptacles are located on the left of the amplifier’s rear panel.

B Power Switch

The power switch is located at the left of the rear panel and is a two position switch. Pushing up the power switch (I is pushed) will power on the device, pushing it down (0 is pushed) will power it off. When the power switch is on the I position, the front panel power LED will be lit.

C Channel 1/2 24VDC battery backup connector

The SPA amplifier can be operated with battery backup. Use the channel 1 and channel 2 “24VDC Battery Backup” connector to do so. These connectors are located on the upper part of the channel connectors on the rear panel.

This 24VDC power supply can be used where no AC power is available. When in use, the front panel Power LED will be lit as well as the front panel DC LED.

D AC Power Inlet

The 3 pin IEC connector is located on the left of the amplifier’s rear panel. It accepts a standard

mains power lead fitted with an IEC connector.

For 115 V, fuse rating:

- SPA 2060: 2.5A
- SPA 2120: 5A
- SPA 2240: 8A

For 230 V, fuse rating:

- SPA 2060: 1.25A
- SPA 2120: 2.5A
- SPA 2240: 5A

E Channel 1/2 Audio Output

For each amplifier channel, there is a dedicated Audio Output connector for connection of either 8 Ohm speakers, 75V speakers or 100V speakers. For the 8 Ohm connection and if you are using more than 1 speaker, please ensure that speakers are wired in a way that the total impedance load is between 8 Ohm and 16 Ohm. The Com point is the – terminal and 8 Ohm, 75V and 100V are the + terminals.

F Fault Contact

The fault contact indicates an amplifier fault by an opening contact.

G Channel 1/2 Outputs

Each channel outputs a replica of the corresponding balanced audio input signal either on XLR or on Euro-terminal block.

H Channel 1/2 Inputs

For each channel, there is an XLR balanced input and a Euro-terminal block balanced input. Each input can be gain adjusted to suit the user’s needs. The On/Off gain switch can instantaneously bypass the gain controller and set the channel gain amplifier to maximum.

4.3.1.4 Characteristics

❖ Case

- Dimension:
  - SPA2060, SPA2120 = 483mm (W) x 305mm(D) x 88mm(H)
  - SPA2240 = 483mm (W) x 420mm(D) x 88mm(H)
- Weight :
  - SPA2060 = 10.6 Kg
  - SPA2120 = 12.3 Kg
  - SPA2240 = 18.5 Kg
- Color = RAL7016.

❖ Power

Item	Voltage	Current Consumption	Comment
AC Input	230V +/- 10%	1.2A	Frequency:47Hz~63Hz
DC Output	21~28V, Typical: 24V	1A	

❖ Audio

- Frequency response = 40 - 20 000 Hz
- Distortion at nominal power: < 1%
- Signal/noise ratio: > 90 dB

❖ Inputs/Outputs

- Power output = 100 V
- Main line input = 770 mV
- Input impedance = 20 KOhms

❖ Current consumption

		Current consumption with 28 VDC supply (per channel)				
		Standby	1/8 Nominal Power (Audi Messages)	1/4 Nominal Power (50% Message, 50% Siren)	1/2 Nominal Power Siren	Nominal Power
Power Consumption	SPA2060	0.19 A	1.5 A	1.8 A	2.75 A	3.2 A
	SPA2120	0.25 A	2.9 A	3.9 A	5.5 A	6.4 A
	SPA2240	0.30 A	5.47 A	7.5 A	10.4 A	12.3 A
	Current consumption with 230 AC supply (per channel)					
	SPA2060	0.07 A	0.32 A	0.42 A	0.53 A	0.67 A
	SPA2120	0.12 A	0.65 A	0.89 A	1.15 A	1.4 A
	SPA2240	0.28 A	1.25 A	1.65 A	2.20 A	2.85 A

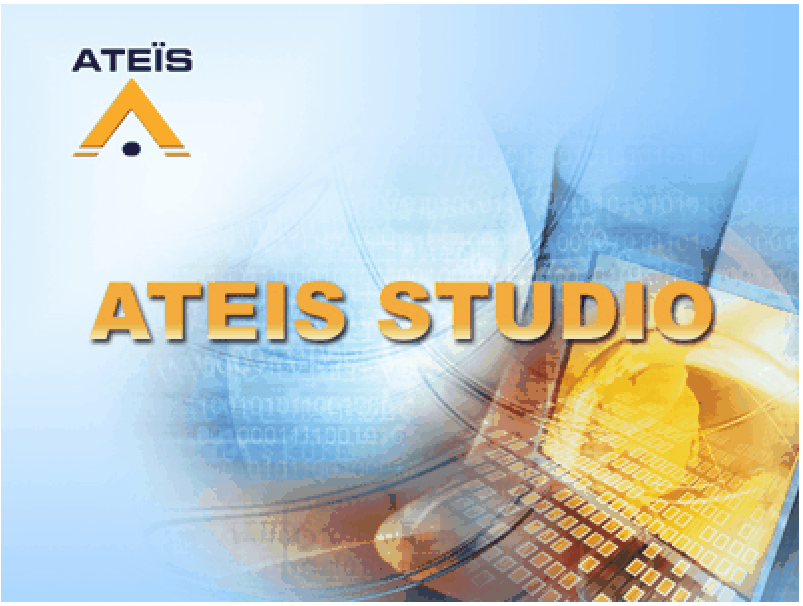
❖ Working Temperature.

- 0°C ~ 40°C

5 System Functionality

5.1 Ateis Studio

5.1.1 Overiew of Ateis Studio



Ateis Studio allows complete audio systems with a multitude of similar or different devices to be configured, supervised and controlled centrally from a single user interface. Ateis Studio supports all IP-based products with the Ateis product family such as IDA8, LAP, ECS. By this way, you can have a large view of the system containing every devices, remote devices and virtual connection between them.

Ateis Studio is ready for future expansion in conjunction with new Ateis products. Ateis Studio also offers control and configuration for power amplifiers, microphone consoles and remote controllers such as PSS AS, URC AS.

It supervises, controls, logs and reports. The whole system configuration can be stored and reloaded at the push of a button (Preset), depending upon the application. Customers can design their own graphic user interface or control panel(s) as well as program automatic sequences (Events) and create different levels of user groups. (Security).

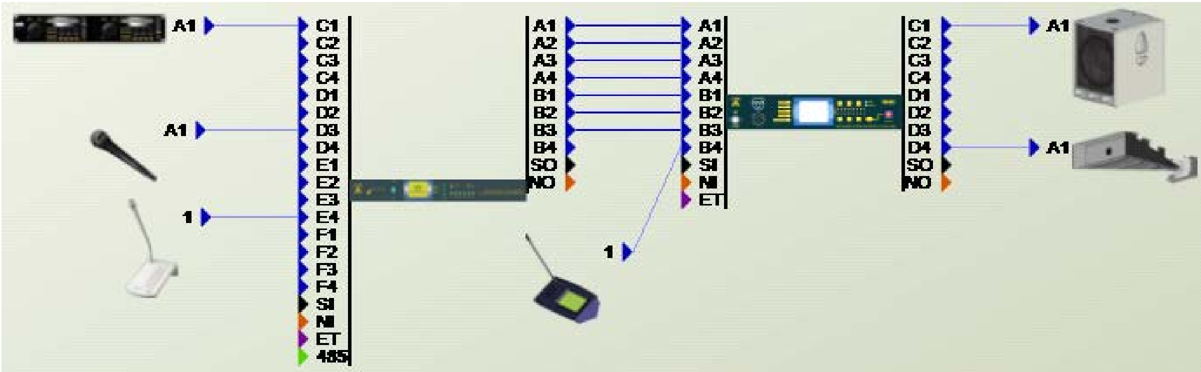
Ateis Studio provides a complete set of tools and building blocks for real-time control, monitoring and design of an audio system or Voice Evacuation system from paging desk to loudspeaker. Detailed information such as signal levels, loudspeaker impedance, pre-recorded messages, amplifier conditions and other parameters can be monitored in real-time.

A library of control and monitoring elements, (GUI) graphical user interface, are provided and includes items such as volume control faders, various metering, high level EQ, Compression, Limiting, Auto-Gain, Noise-sensing, Mixing, quick-buttons and display elements. The behaviour of these elements are completely configurable for customer control design.

The combination of these GUI tools allows a user to create a control surface that is effective, easy to operate regardless of the user’s technical knowledge. Additional security is available through the use of password protected layers according to the EN 54-16. Multiple users can be created within Ateis Studio, each with a unique password and access to specific layers of the GUI. This creates a control surface that is specific to the needs of any system designer and/or operators.

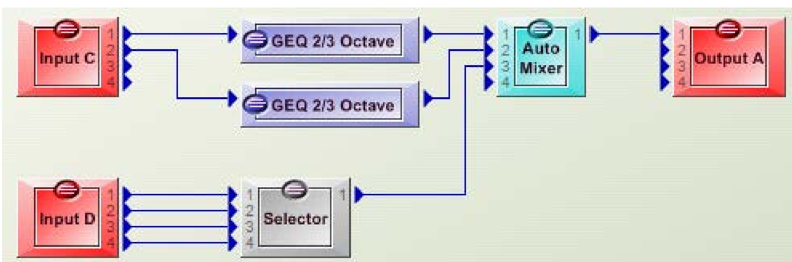
In addition you will find a comprehensive selection of tools for system design and programming using our Ateis Studio integrated software platform for all remote-controllable devices and systems.

As you can have a huge system, the software provides a lot of function making the navigation inside the system easier by using different layers of the design, freely configurable.



After the determination of the devices included in your system, you can start designing the internal signal paths independently for each device, with all the exiting features the huge component library offers, simply by drag and drop and connect different audio component.

This all is done in a straight forward and clear manner, helping you to concentrate on creating state-of-the-art electro-acoustic environments.



This software allows management of all devices in the network from one computer. That means from one computer and in one software interface you are able to update all the devices, to modify the configuration and to store it into each devices independently, and of course to check and update them.

After that the design is successfully compiled and loaded into the machine(s), you have full access to all the parameters in the system.

There are a lot of controlling features starting from simple TTL I/O's over analogue voltage control to advanced data communication. You can choose from a range of controlling devices or remote microphones suitable to access parameters within the whole networked system.

5.1.2 System Requirement

❖Ateïs Studio runs on window XP, Vista and Seven.

❖We recommend at least 1GB RAM, 2GHz processor.

❖Disk Space required is 100MB.

5.1.3 Install Ateis Studio

Please follow the steps below to install Ateis Studio:

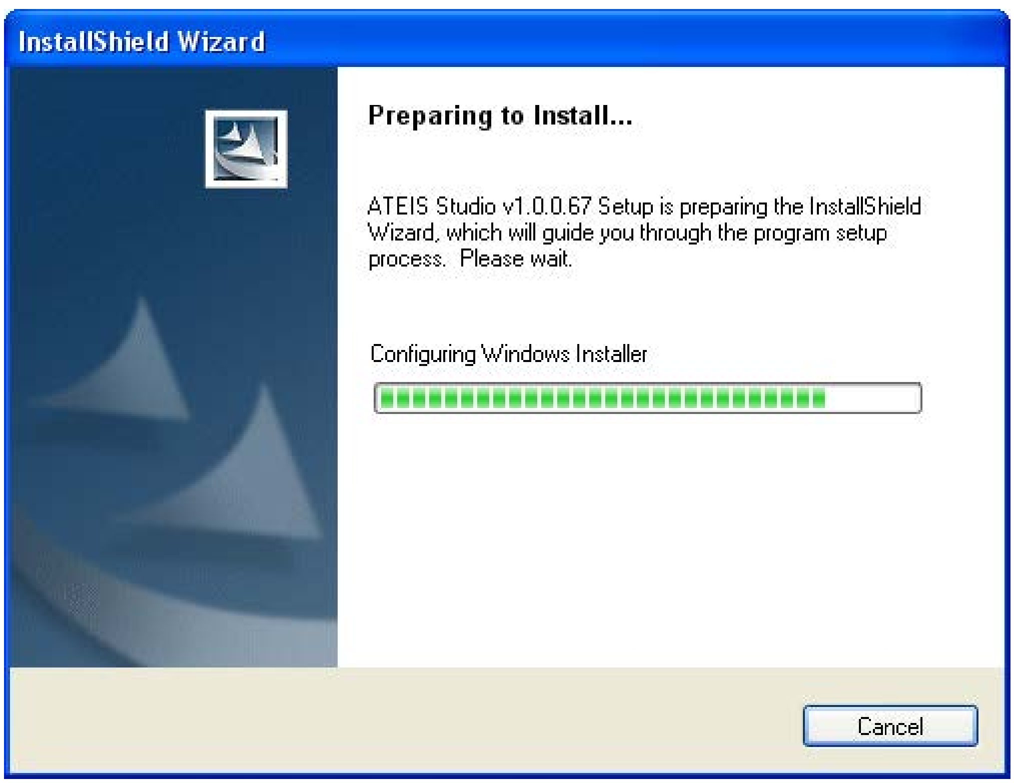
1. Before installing the Ateïs Studio software, check if your PC meets the system requirements.

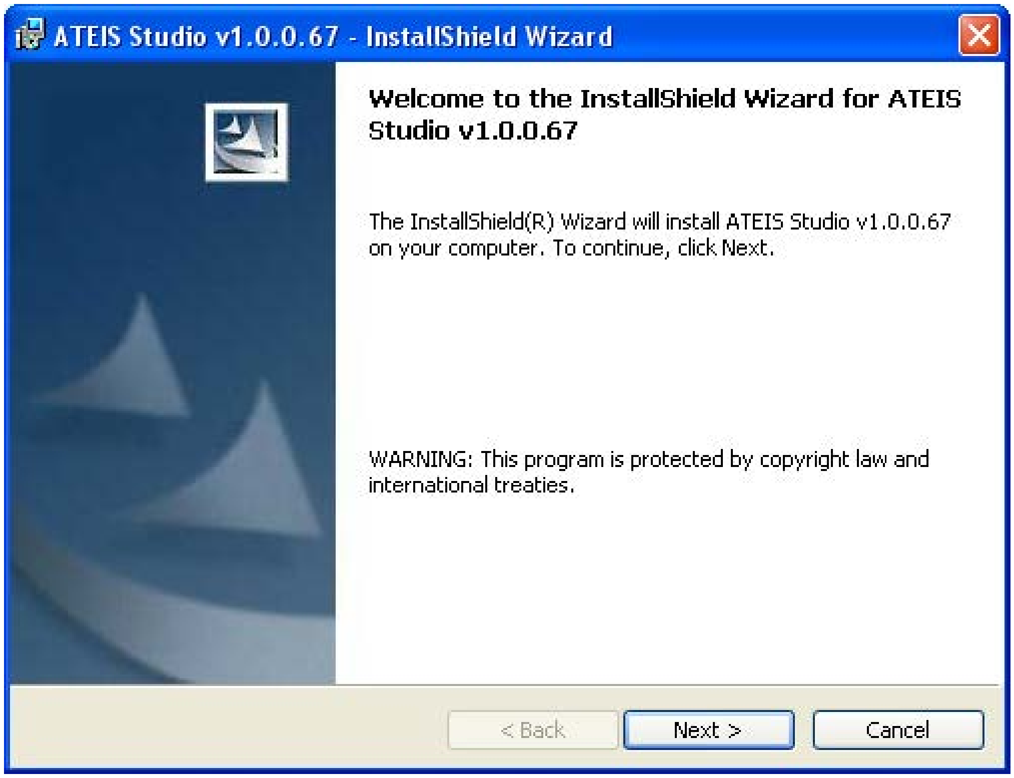
2. Turn on your PC and allow Windows to start up. Insert the supplied CD-ROM into your drive. If the content of the CD-ROM is not displayed automatically double click the 'My Computer' icon and navigate to the 'DVD/CD' drive.

Note: If you have access to the WEB: goes to www.ateis-international.com and download the latest software version. You may need to update the firmware.

3. Copy the file 'ATEIS Studio v.1.x.x setup' using the copy and paste commands or by drag and drop to the 'Desktop'. Wait until the task has finished.

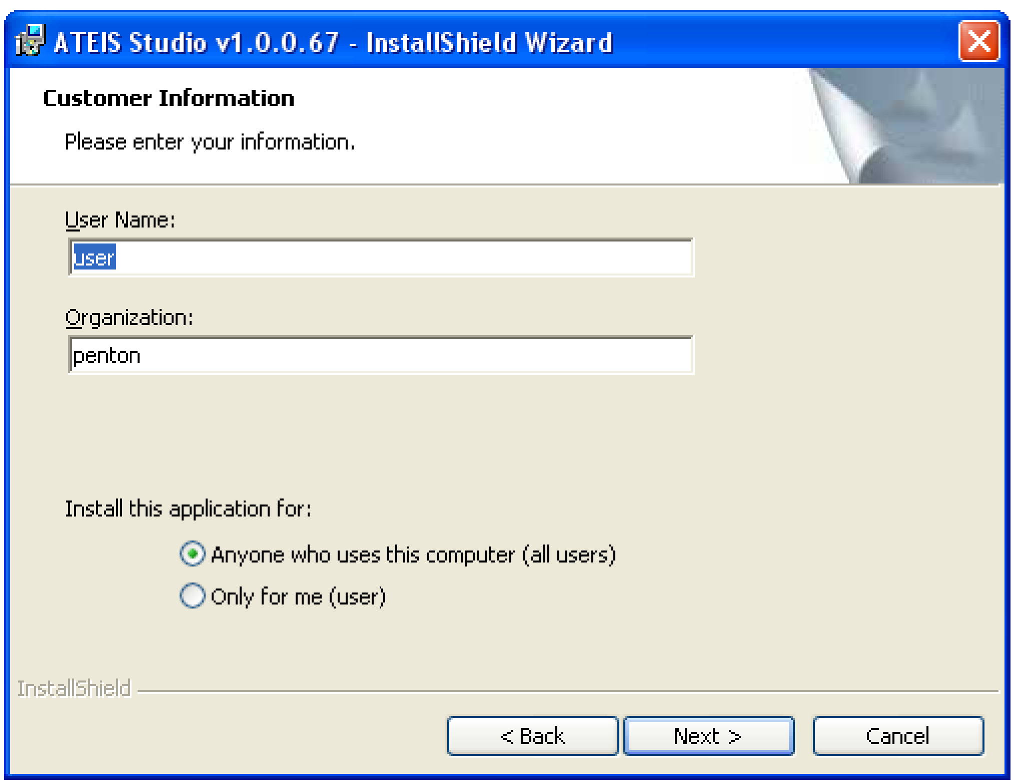
4. Start the program 'ATEIS Studio vx.x.x.x setup' by double click on its icon and follow the instructions to install the software onto your hard drive.



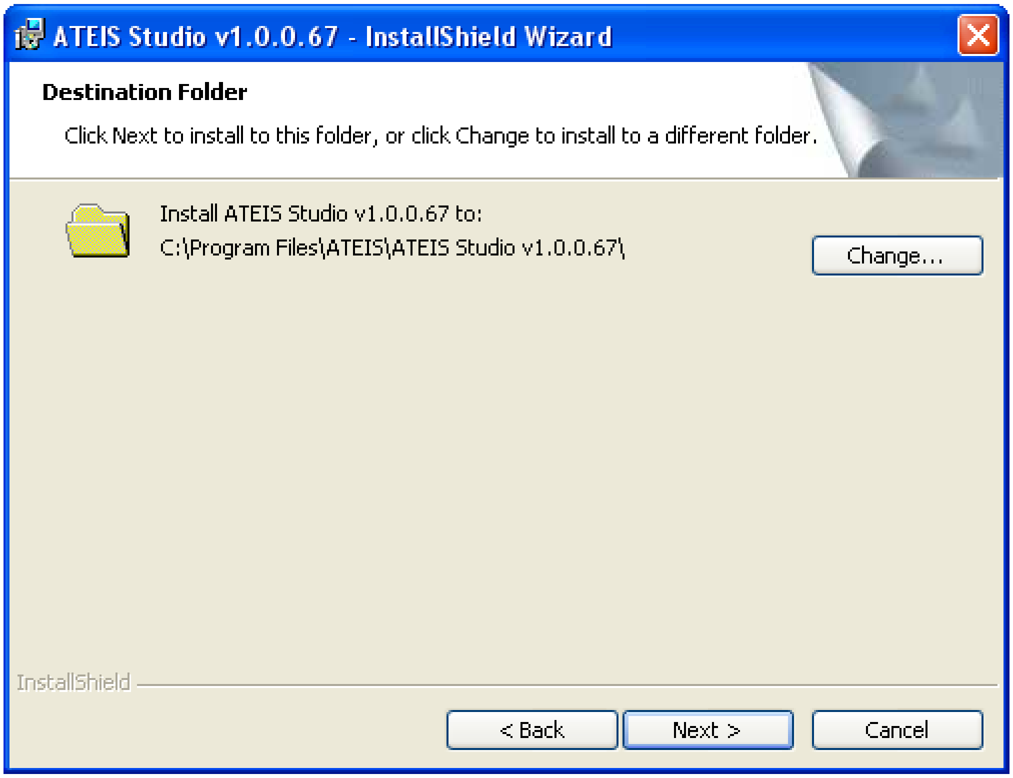


Click [Next]

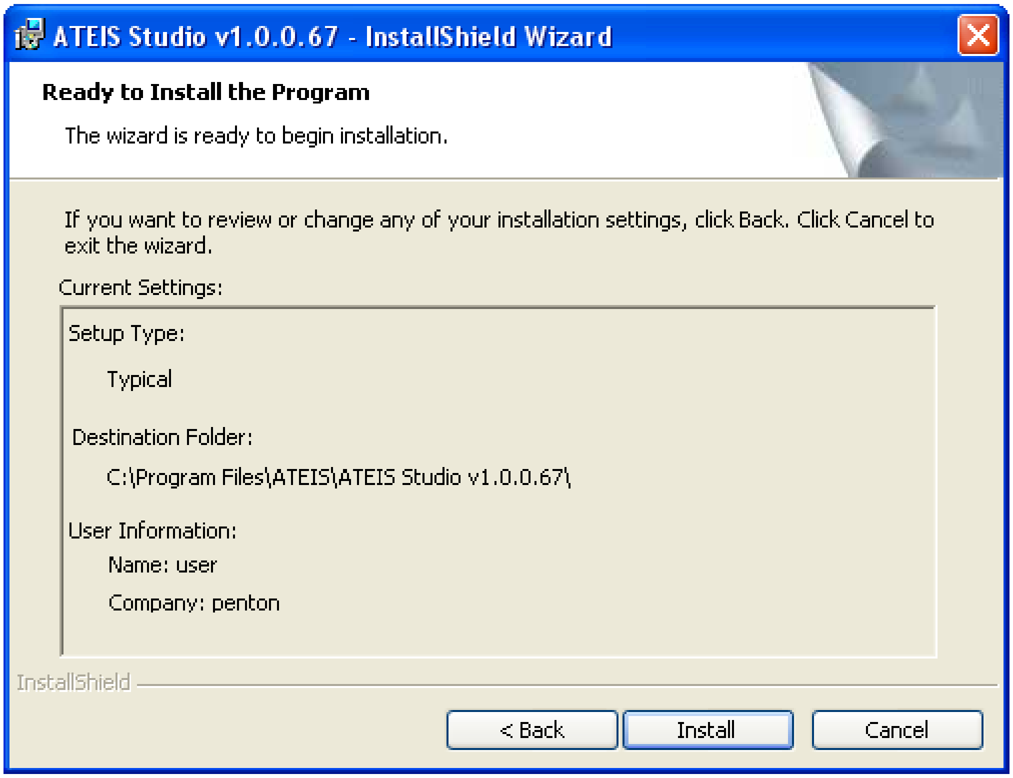
5. Type in user name and your organization name if you want and click [Next]



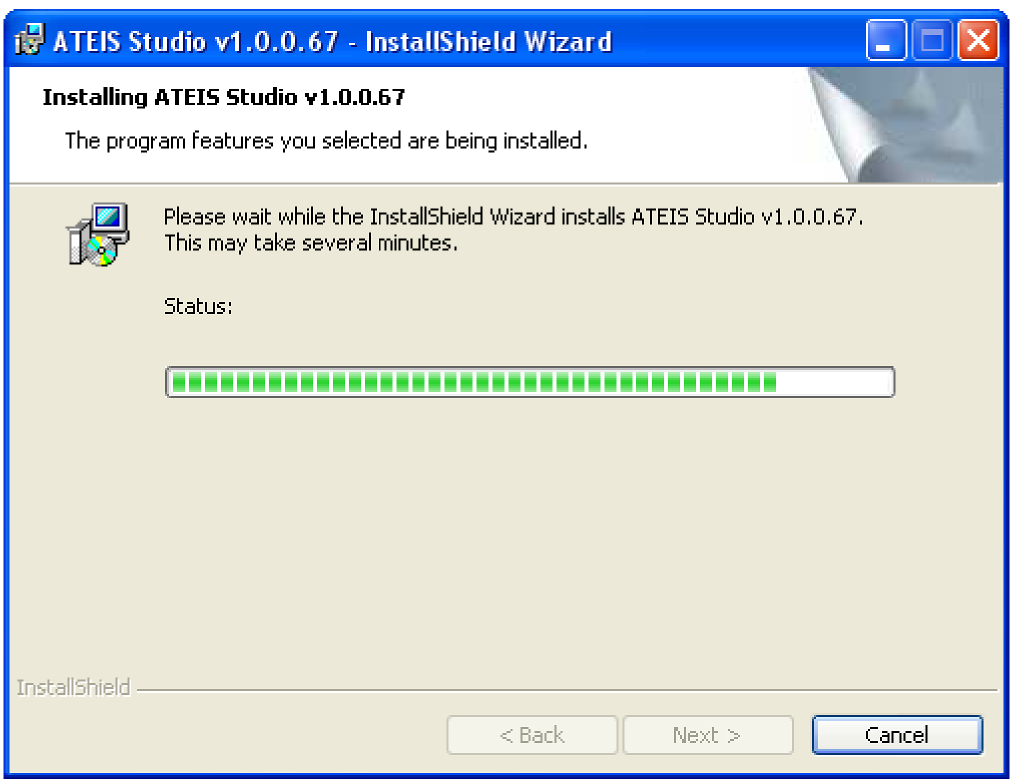
6. By default the program the software is installed under the path "C:\Program Files\ATEIS\ATEIS Studio vx.x.x.x". Change installation path by clicking [Change...] if required, and click [Next].



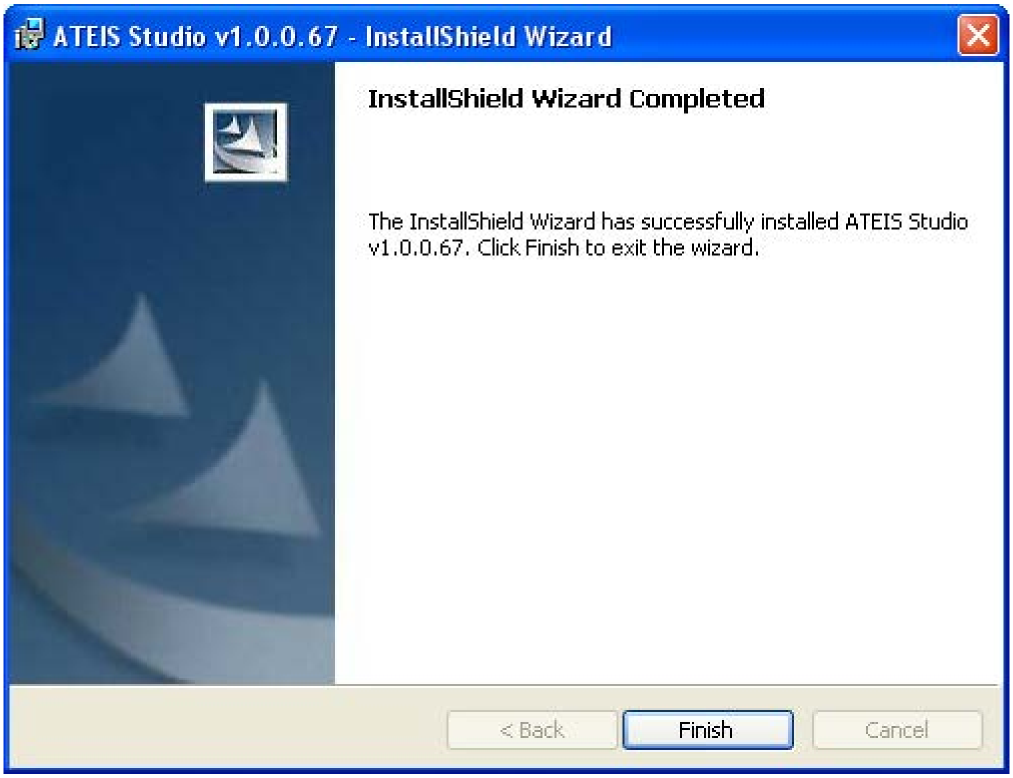
7. As usual with the install shield wizard a window appears showing all information before the actual installation. You can enter corrections at this point by clicking [< Back]. If there is nothing to change click [Next >] to execute the installation.



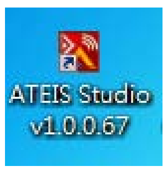
8. Wait until the end of installation.



9. After all installation tasks have been finished successfully click [Finish] to exit wizard.

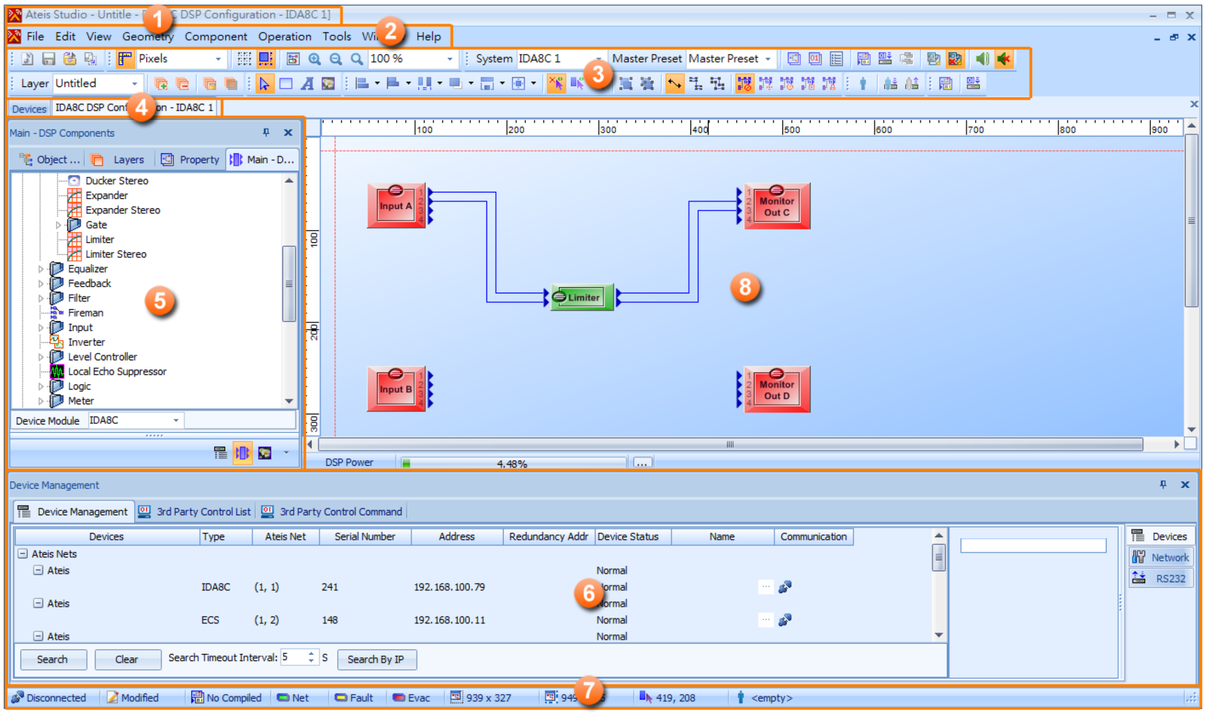


10. The installation program creates a shortcut to the "ATEIS Studio v.1.x.x .exe" named "ATEIS Studio vx.x.x.x" on your "Desktop"'



Double click the icon to start the program.

5.1.4 Windows Layout



The basic windows layout should like the figure showed above, but the layout can be changed if you drag the dock panel(circle No.5 & 6 in are examples of dock panel). For each part of area in the software which is labeled using orange circles in the figure, there are brief description of them:

1. Title Bar

   Displays the path name of the current file.

2. Menu Bar

   Holds all the functionalities of the software (windows and tools), arranged by topic.

3. Tools Bar

   Short access to main functionalities that are hidden into menu bar.

4. Tab Bar

   Shows the window title of all opened windows (design, components control windows) for quick access.

5. Dock Panel-Component Template

   A dock panel on left side of software, it contains all devices and DSP components you can use to build up and design your system.

6. Dock Panel-Device Management

   A dock panel on bottom side of software, it list information about devices and allow user to connect to device for maintenance.

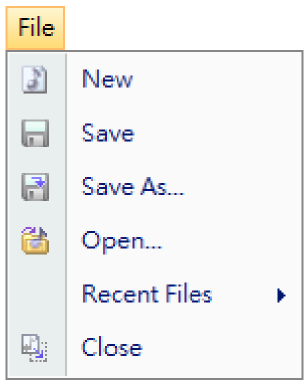
7. Status Bar

   Display the status of software, including the modification indication of editing file, mouse cursor position, ... etc.

8. Working Area-DSP Components Editor

   A graphical interface allow user to design system, put the components which performs specific audio signal processing, and wiring between components to make complete audio path from input to output.

5.1.5 File



- New

  Opens a new blank project.

- Save

  Saves the current file you are currently working with.

- Save As

  Opens the “Save as…” dialog box to save the current working project with a different name and/or path.

- Open

  Opens the “Open...” dialog box which allows you to select a path and open a saved file.

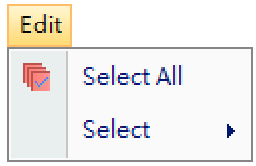
- Recent Files

  Quick access to the last opened file.

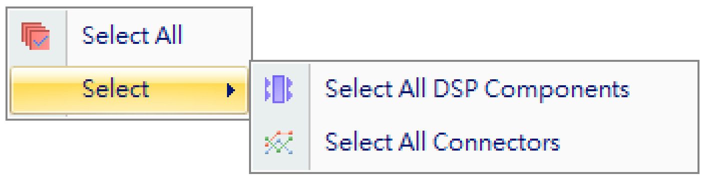
- Close

  Close the opened files.

5.1.6 Edit

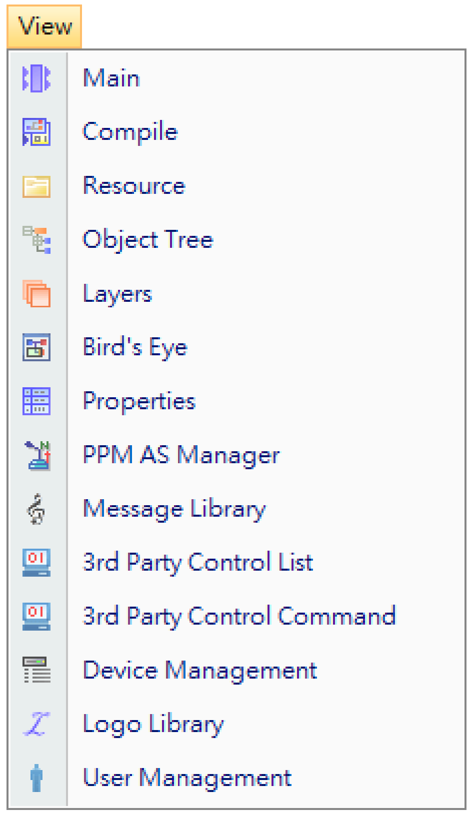


- Select All 
    - Select all the components and wires.
- Select



- Select All DSP Components 
    - Select all DSP components in the graphic editor.
- Select All Connectors 
    - Select all wires in the graphic editor.

5.1.7 View



Each item in the [View] menu can open a specific window to do settings modification or management of devices.

- Main

  Open the window of Component Template which contains various type of devices and DSP components for designing the system.

- Compile

  Open the compiler window and compile the current configuration.

- Resource

  Open the window of Resource Manager to manage resource files in the software.

- Object Tree

  Open the window of Object Tree to see the whole structure of the configuration file.

- Layers

  Open the window of Layer Manager to manage the graphical layers for objects.

- Bird's Eye

  Open the window of Bird's Eye to have a overview of graphical objects.

- Properties

  Open the window of Properties which allow you to inspect various parameters of an object in graphic editor.

- PPM AS Manager

  Open the window of PPM AS Manager to have a overview of all PPM AS in configuration file.

- Message Library

  Open the window of Message Library to manage message files in the software.

- 3rd Party Control List

  Open the window of 3rd Party Control List to have a overview of elements that are 3rd party controllable.

- 3rd Party Control Command

  Open the window of 3rd Party Control Command that display 3rd party control string of focused element.

- Device Management

  Open the window of Device Management that discovery the devices over network and maintenance.

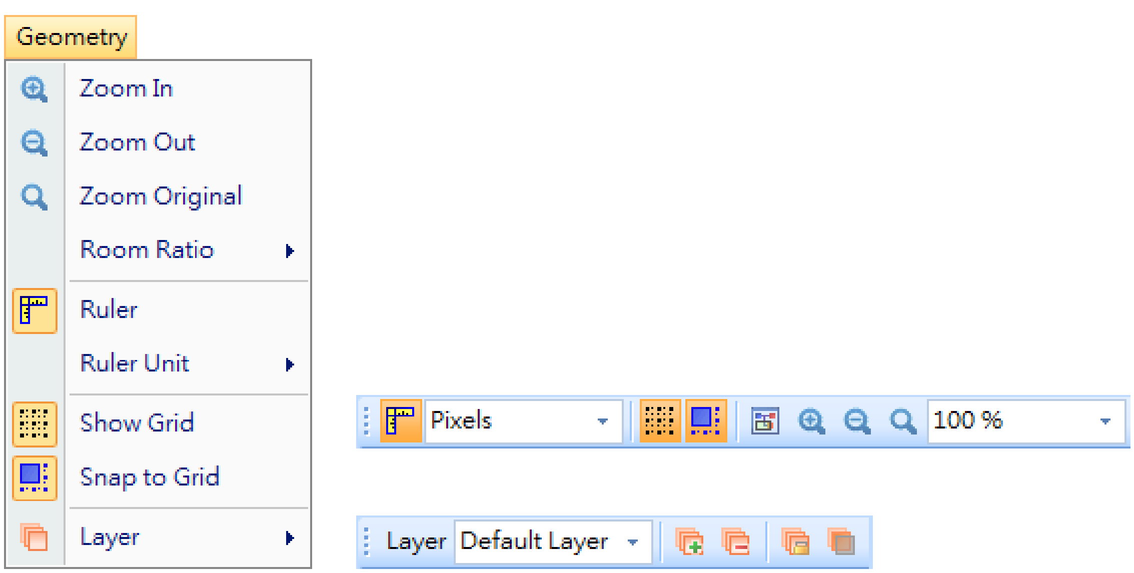
- Logo Library

  Open the window of Log Library to manage Logos for Ateis devices.

- User Management

  Open the window of User Management for editing the users of configuration file.

5.1.8 Geometry



- Zoom In

Zoom in the graphic editor.

- Zoom Out

  Zoom out the graphic editor.

- Zoom Original

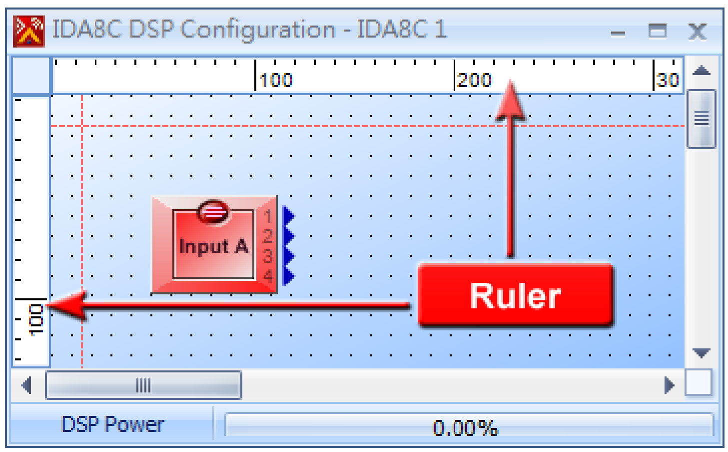
  Zoom to original size of graphic editor.

- Room Ratio

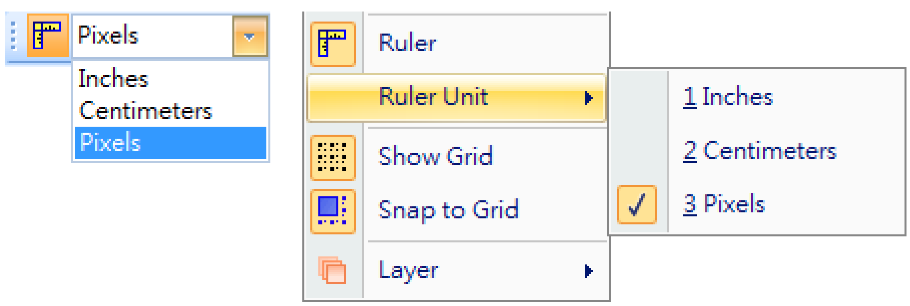
  Set graphic editor to a specific ratio.

- Ruler

  To show or hide ruler of graphic editor.



- Ruler Unit



  Specify ruler unit of graphic editor, there are three types of unit:

  - Inches
  - Centimeters
  - Pixels

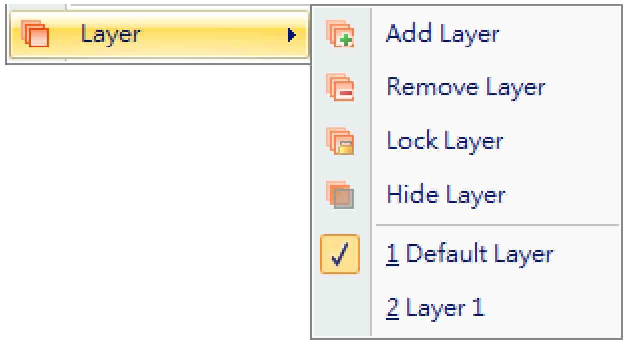
- Show Grid

  To show or hide grid of graphic editor.

- Snap to Grid

To enable or disable moving an object by multiple pixels at a time.

- Layer



  - Add Layer

    Add a layer of graphic editor.

  - Remove Layer

    Remove a layer of graphic editor.

  - Lock Layer

    Lock a layer of graphic editor to disable moving of any objects belong that layer.

  - Hide Layer

    Hide a layer, all objects belong that layer will become invisible.

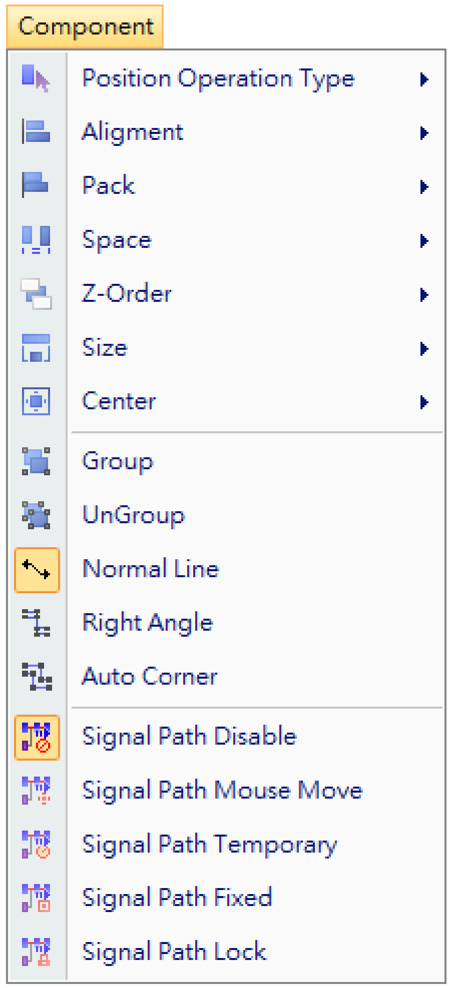
  - Select Layer

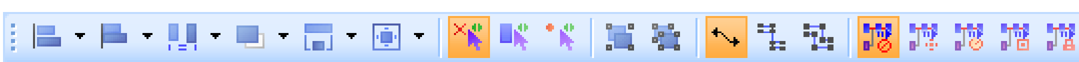
    Select a layer to be current, any newly created object will belong to current layer.

- Bird's Eye

  Open the window of Bird's Eye that allow you to see an overview of graphic editor.

5.1.9 Comoponent Editing





❖Position Operation

- Position Operation Type 

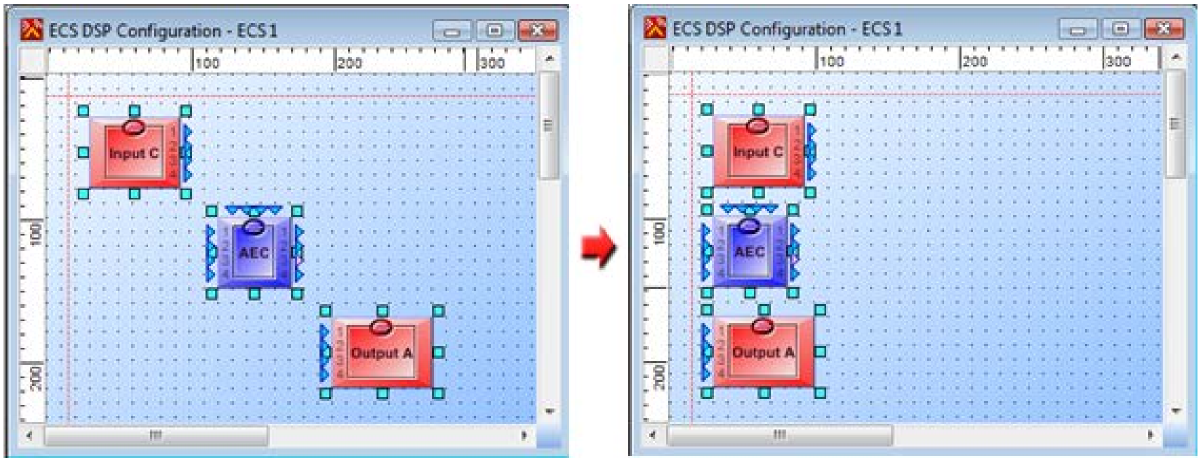
  Some functions, as Alignment or Pack use a reference. Here you define what kind of reference you use. It can be the place in the area (Normal), a component or a point.

  For example when you will use the "Align Left" function to align several components to the left, the left will depend of the reference.

  - Normal 

    In Normal mode, the reference is the place in the graphic editor.

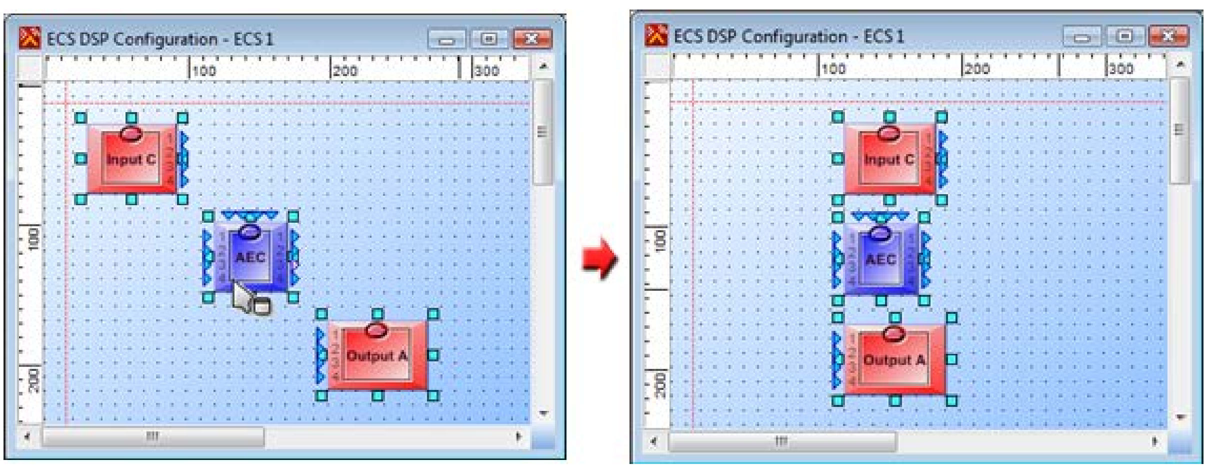
    If you select several components, and use the "Align Left" function, then the reference will be the left side of component at the left of the area. All the selected components will be aligned to this reference.



- Reference Component

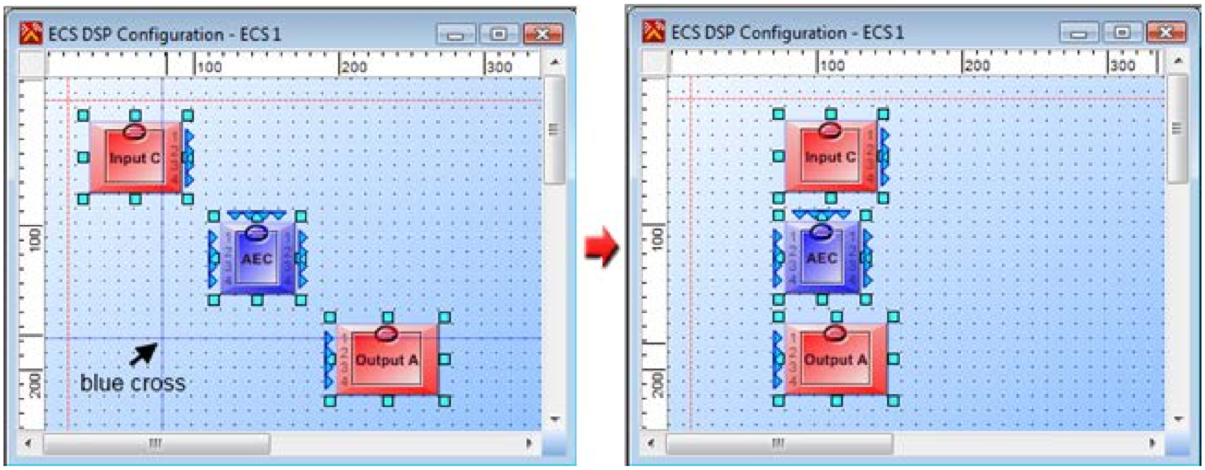
  In Reference Component mode, the reference is a component. If you select several components, and use the "Align Left" function, then the mouse cursor will change to , asking you to select a component as reference. Select a component by clicking in. The mouse change to when it is placed on a component. All the selected components will be aligned to this component.

  In this example, we choose the AEC component as reference.

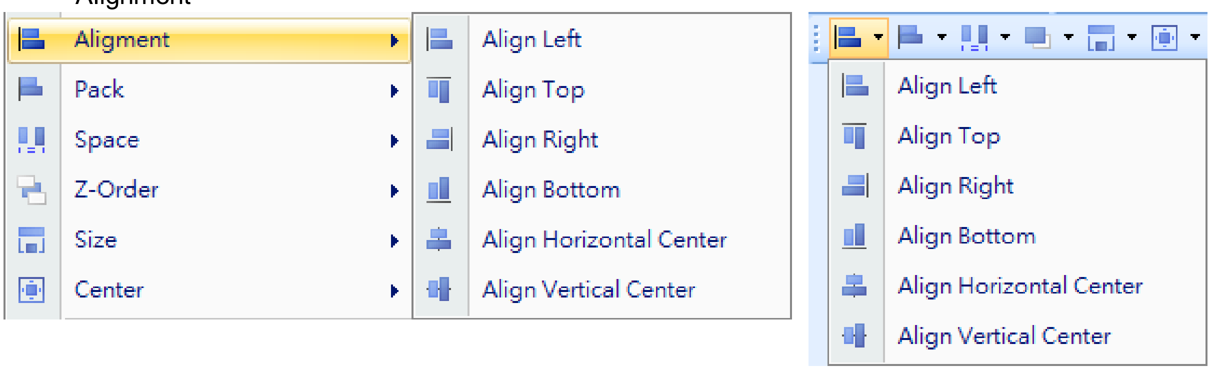


- Reference Point

  In Reference Component mode, the reference is a point. If you select several components, and use the "Align Left" function, then the mouse cursor will change to a blue cross, asking you to choose exactly where is the reference. All the selected components will be aligned to this point.

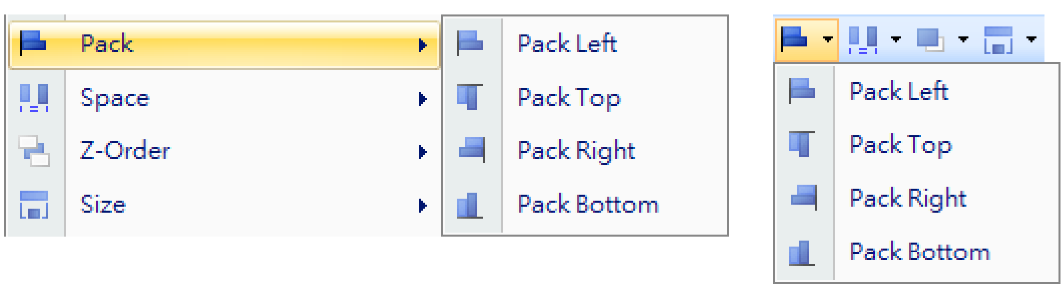


- Alignment



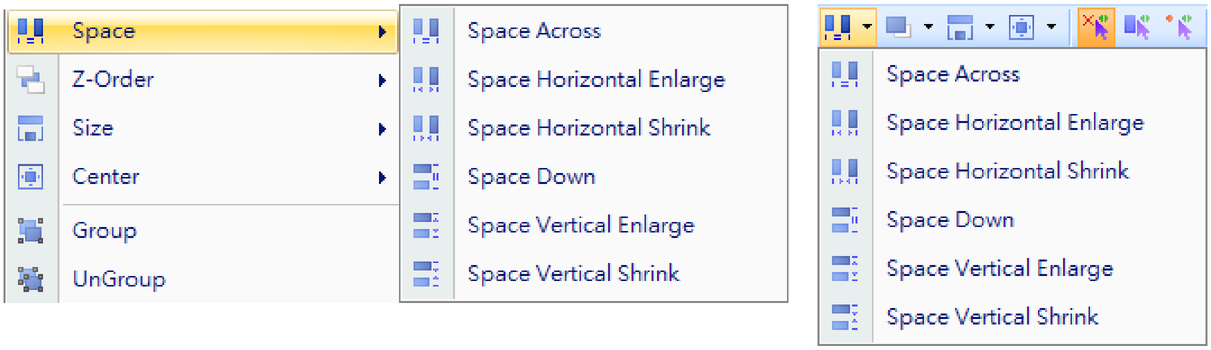
You can select several elements (by maintaining left click and move the mouse you will have a selection square) then you can align all the elements. This function uses a reference which depends of the reference mode

- Pack



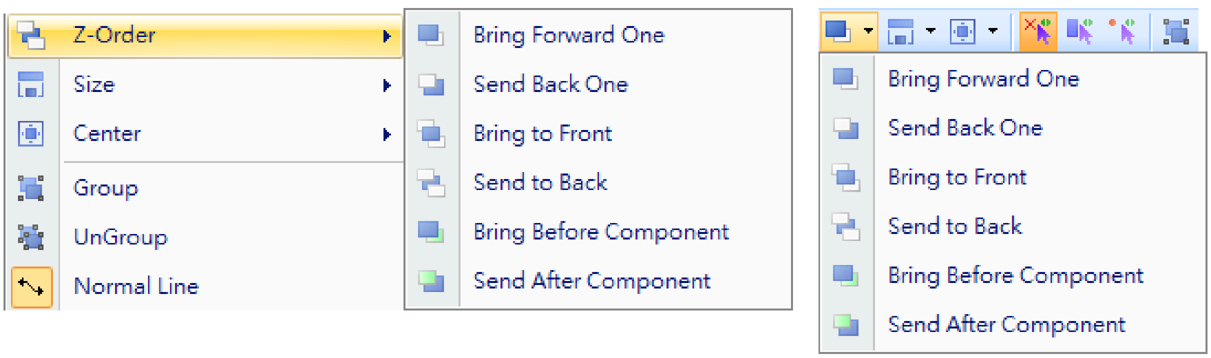
You can select several elements (by maintaining left click and move the mouse you will have a selection square) then you can pack the elements. This function uses a reference which depends of the reference mode.

- Space



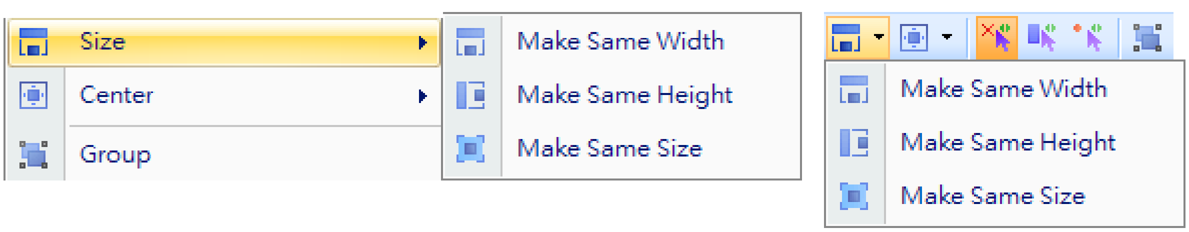
You can select several elements (by maintaining left click and move the mouse you will have a selection square) then you can define the space between all the elements. It is interesting if you pack the elements first.

- Z-Order



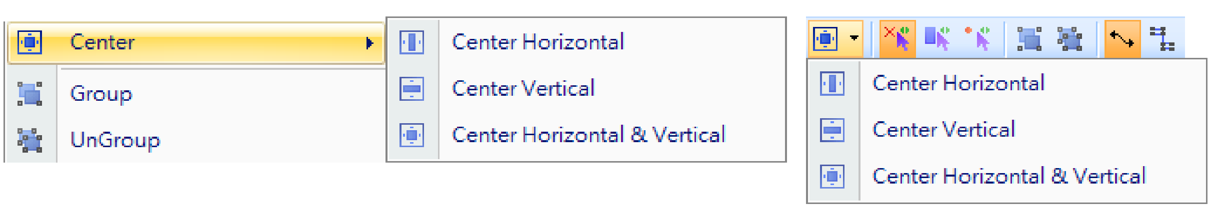
If you have elements which are under other elements, you can define by selecting one element if this will be to the front or to the back. Pay attention sometimes element can be totally hidden with this action. It is useful with background image and buttons for example.

- Size



These functions resize the component. You can select several components and adjust them to have the same dimensions. All the selected component will have the size of the component on the window's top-left.

- Center

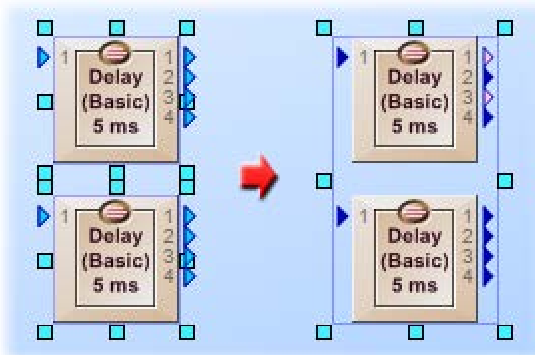


You can select several component (by maintaining left click and move the mouse you will have a selection square) then you can move the component to the window's middle, horizontally or vertically.

❖ Grouping

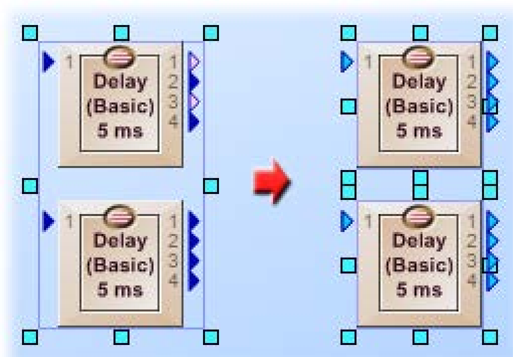
- Group 

  You can select several component (by maintaining left click and move the mouse you will have a selection square) then you can group the components in one component.



- Ungroup 

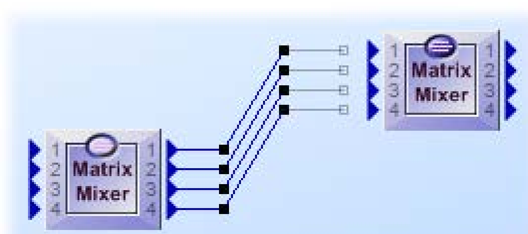
  This function split grouped components to single component. It's the reverse of the Group function.



❖ Wiring Mode

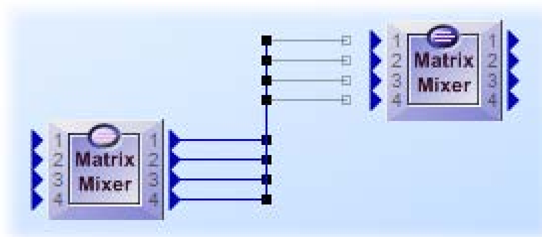
- Normal Line 

  Select Normal Line if you want to draw the wire with different angles.



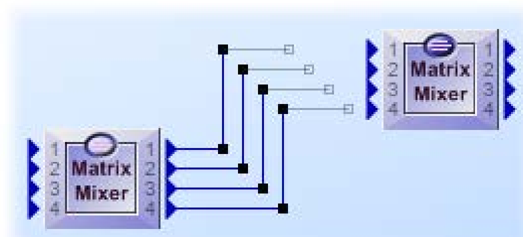
- Right Angle

Select Right Angle if you want to draw the wire only with right angles (90 degrees).



- Auto Corner

Select Auto Corner if you want that the wires are automatically arranged to have proper corner, with separated wires and with right angles.



❖ Signal Path Mode

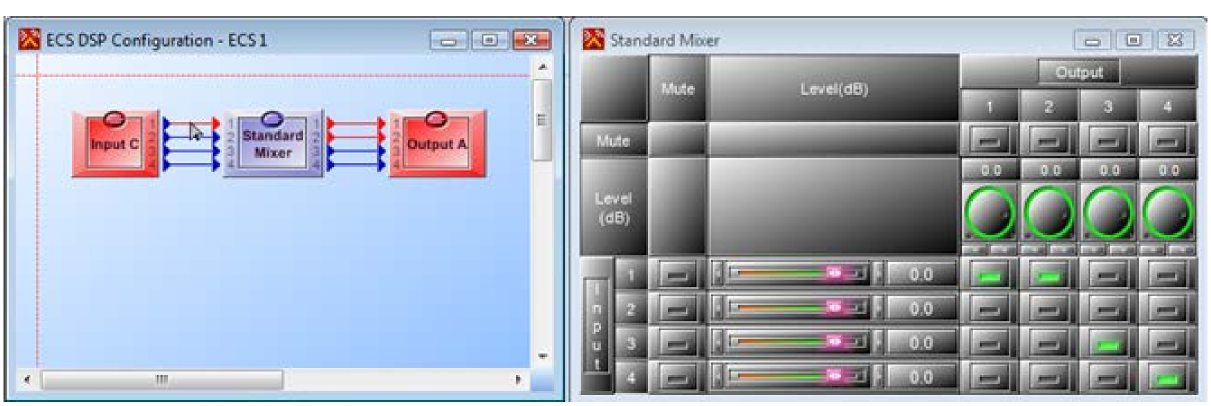
- Signal Path Disable

Disable the signal path highlighting function. When it's not disabled, the signal path function allows seeing where an audio signal is sent through the design. The signal path is highlighted in red. To understand what this function is, please see the next chapters.

- Signal Path Mouse Move

In this mode, by placing the mouse on a signal link, the signal link and the entire signal path linked will be highlighted in red.

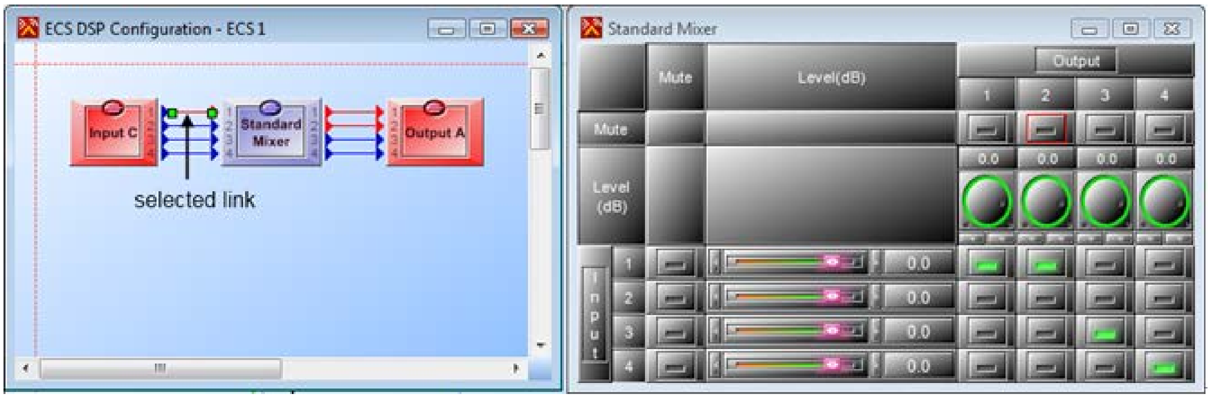
In this example the input C1 is routed in the matrix to the output 1 and 2. If you place the mouse on the link of the channel 1, then the signal path on the output 1 and 2 will be displayed in red.



- Signal Path Temporary

In this mode, by selecting a signal link, the signal link and the entire signal path linked will be highlighted in red. If you click in a free zone, the signal path is cancelled.

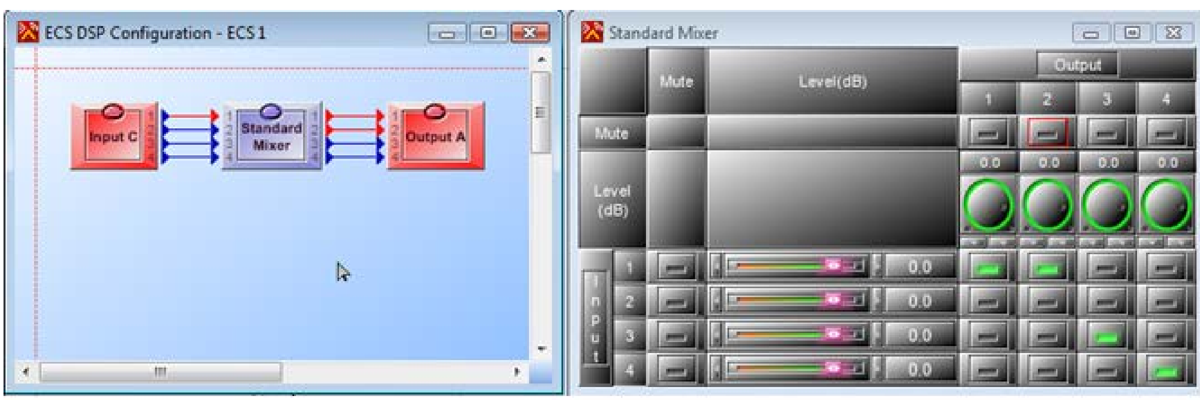
In this example the input C1 is routed in the matrix to the output 1 and 2. If you place select the link of the channel 1, then the signal path on the output 1 and 2 will be displayed in red.



- Signal Path Path Fixed

In this mode, by selecting a signal link, the signal link and the entire signal path linked will be highlighted in red. The signal path is fixed, to cancel it you have to click on another link or component (not on a free zone).

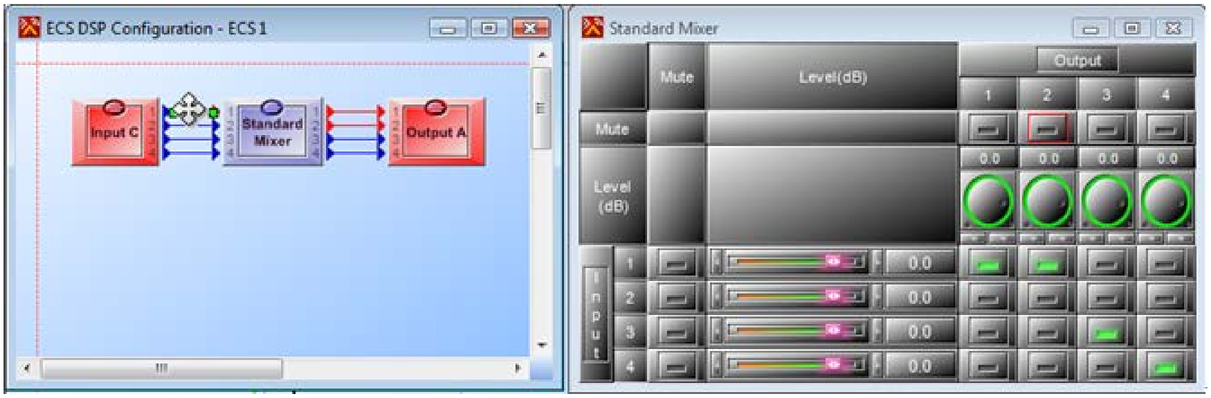
In this example the input C1 is routed in the matrix to the output 1 and 2. If you place select the link of the channel 1, then the signal path on the output 1 and 2 will be displayed in red.



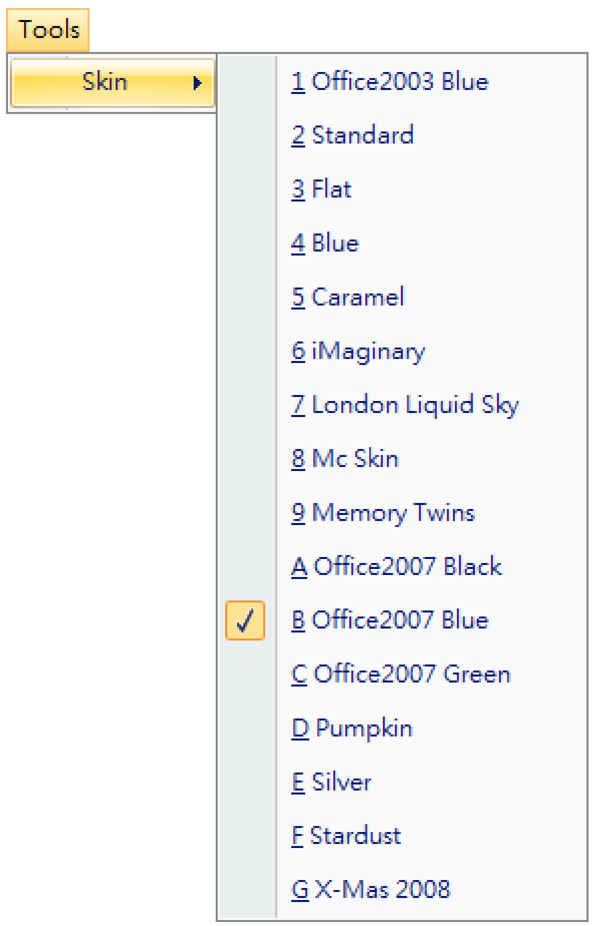
- Signal Path Lock

In this mode, by selecting a signal link, the signal link and the entire signal path linked will be highlighted in red, then the mouse cursor will change into , asking you to select the link again, in order to lock it. The signal path is locked, to cancel it you have to select the mode "Signal Path Disable ".

In this example the input C1 is routed in the matrix to the output 1 and 2. If you place select the link of the channel 1, then the signal path on the output 1 and 2 will be displayed in red.



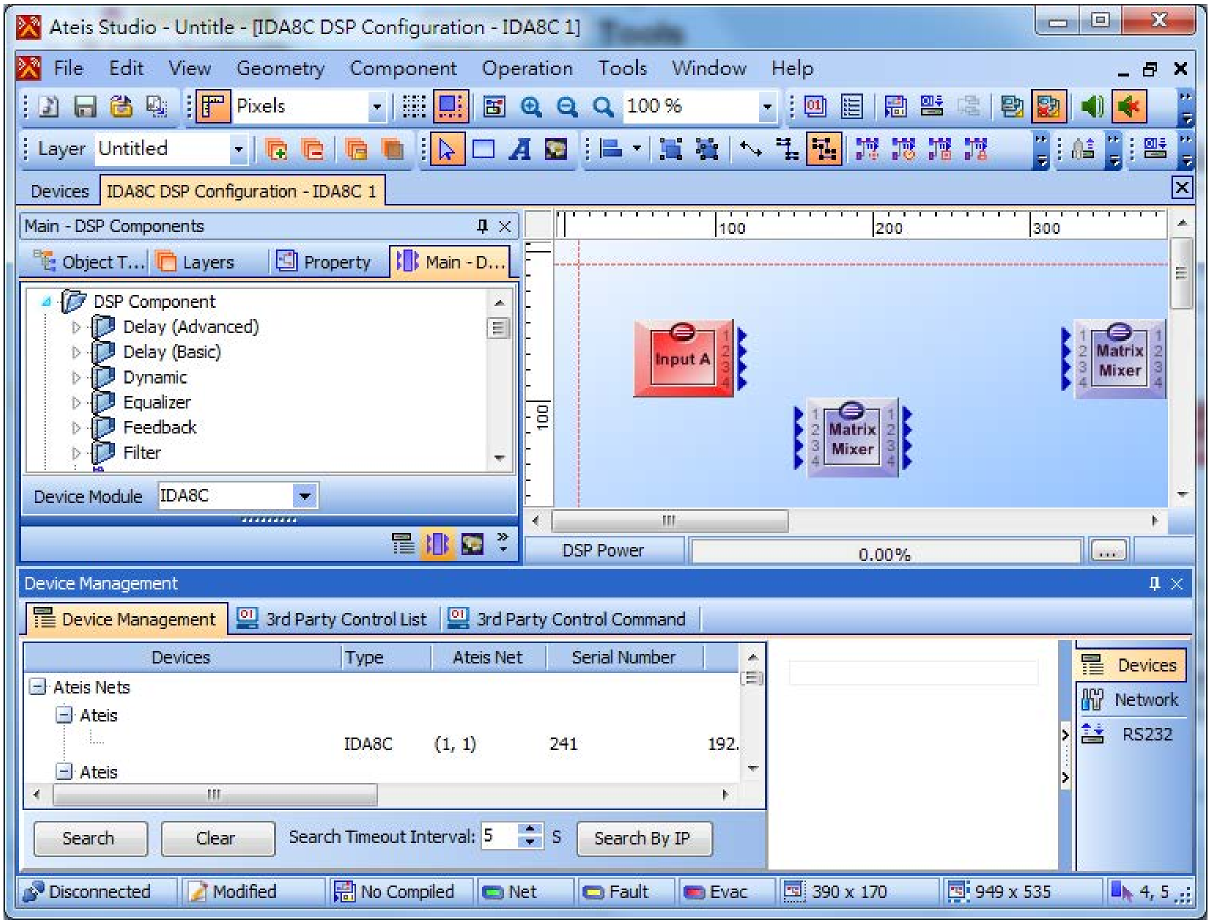
5.1.10 Tools



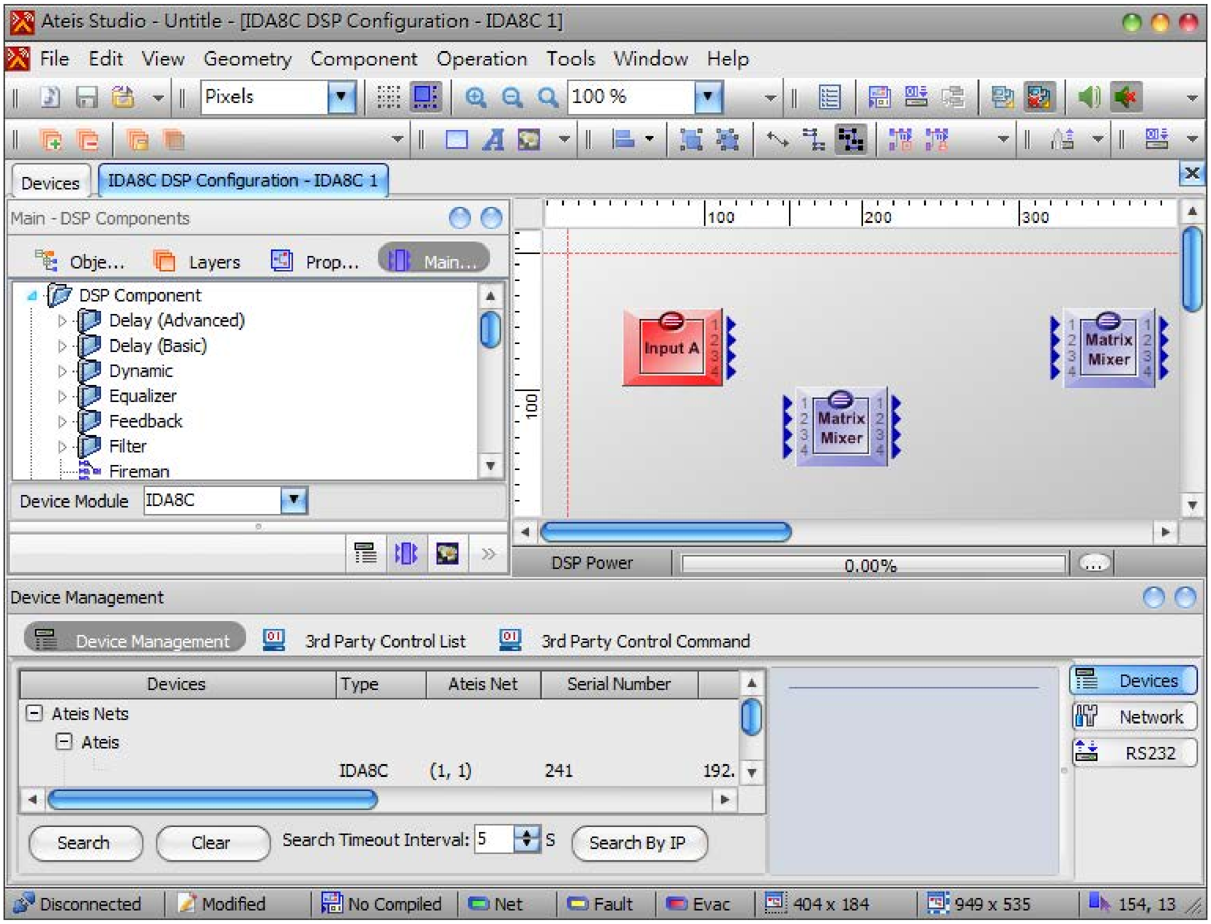
By changing skin you modify the global style (colour and appearance) of the windows. You can choose one of the skin in the long list.

Here are examples of skins:

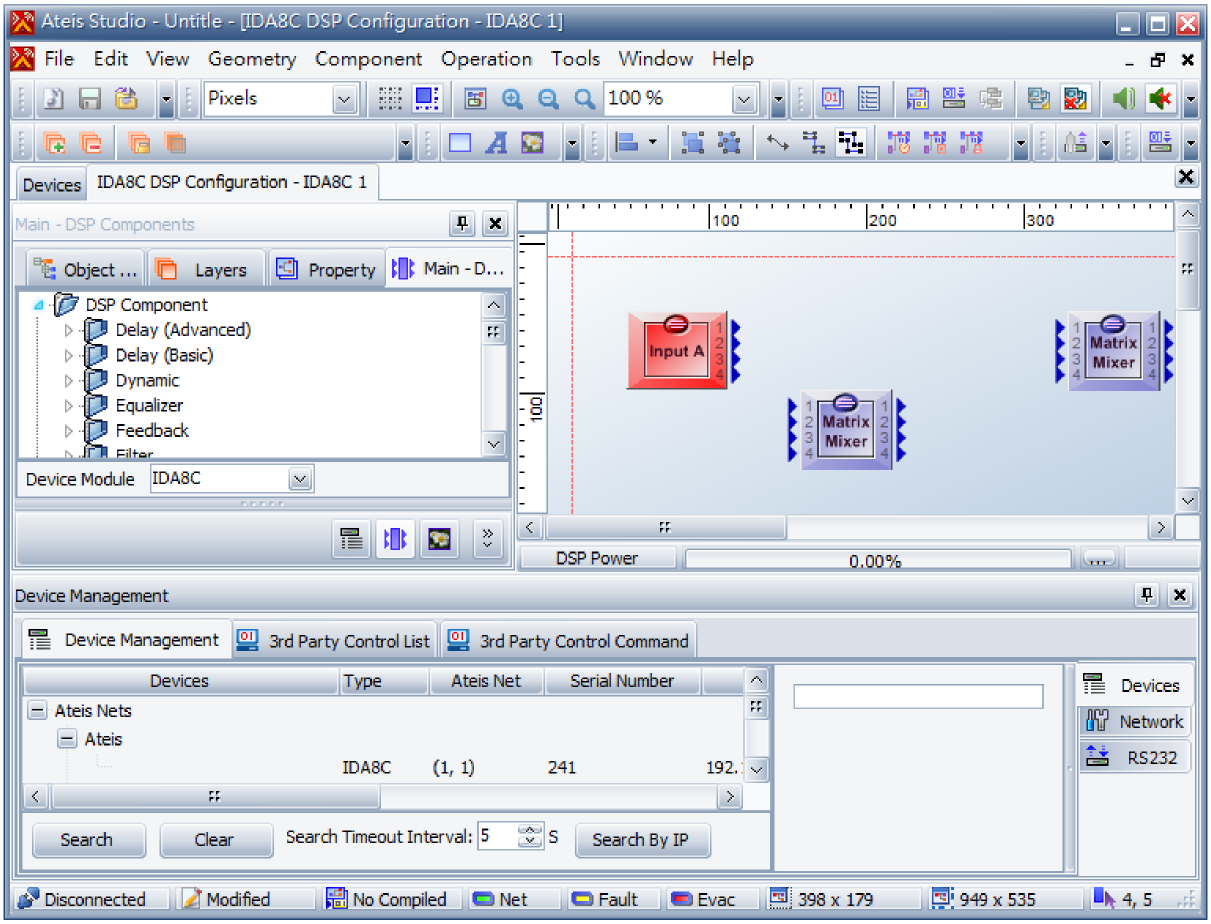
- Office 2003 Blue


Ateis Studio - Untitle - [IDA8C DSP Configuration - IDA8C 1]
File Edit View Geometry Component Operation Tools Window Help
Pixels
100 %
Layer Untitled
Devices IDA8C DSP Configuration - IDA8C 1
Main - DSP Components
Object T... Layers Property Main - D...
DSP Component
Delay (Advanced)
Delay (Basic)
Dynamic
Equalizer
Feedback
Filter
Device Module IDA8C
Input A
Matrix Mixer
Matrix Mixer
DSP Power
0.00%
Device Management
Device Management 3rd Party Control List 3rd Party Control Command
Devices Type Ateis Net Serial Number
Ateis Nets
Ateis
IDA8C (1, 1) 241 192.
Ateis
Devices
Network
RS232
Search Clear Search Timeout Interval: 5 S Search By IP
Disconnected Modified No Compiled Net Fault Evac 390 x 170 949 x 535 4, 5

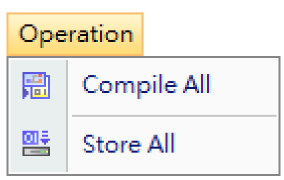
- Mc Skin

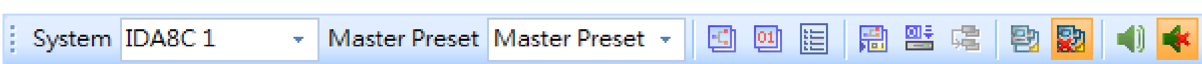


- London Liquid Sky



5.1.11 Operation





In this menu you will find the major operations related to system connection, allowing to compile the configuration design, and then to store the compiled configuration into the device.

Most part of those windows has short access in the Tools bar. We are going to introduce you every items. To learn how to use those items please refer to the “create a project” and “how to” sections.

❖System Selector

The combo box close to text "System" to switch system to be current system, all actions in the tools bar is working on the current system. The actions include master preset changing, store, online/offline, audio enable/disable.

❖ Master Preset Selector

The combo box close to text "Master Preset" to switch master preset.

❖ Master Preset Settings

Open the master preset settings window.

❖ Sub-Preset Settings

Open the sub-preset settings window.

❖ Event

Open event settings window.

❖ Compile All

Start the compile process. The software verifies that no major errors remain before uploading the design into Audio Processor.

❖ Store All

Start the compilation process and then uploads your design into the audio processor.

❖ Save Parameters

Save parameters to device's flash.

❖ Online

To make Ateis Studio online with audio processors, after online, the parameter will be synchronized, adjustment of parameters too. When Ateis Studio is online with devices, designing configuration is not allowed.

❖ Offline

To make Ateis Studio working along. In offline state, you can modify the configuration file.

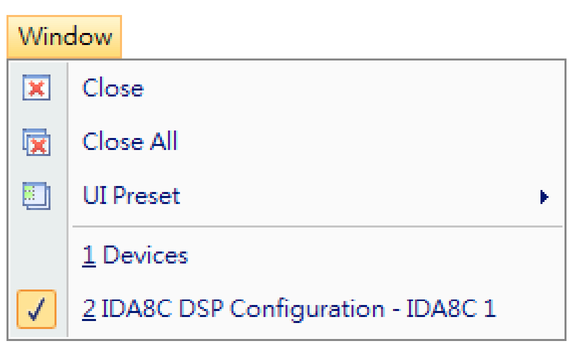
❖ Audio Enable

Enable audio of ateis devices.

❖ Audio Disable

Disable audio of ateis devices.

5.1.12 Window



In this topic you will find tools to re-arrange windows or find hidden opened windows.

❖ Close

Close the opened active window.

❖ Close All

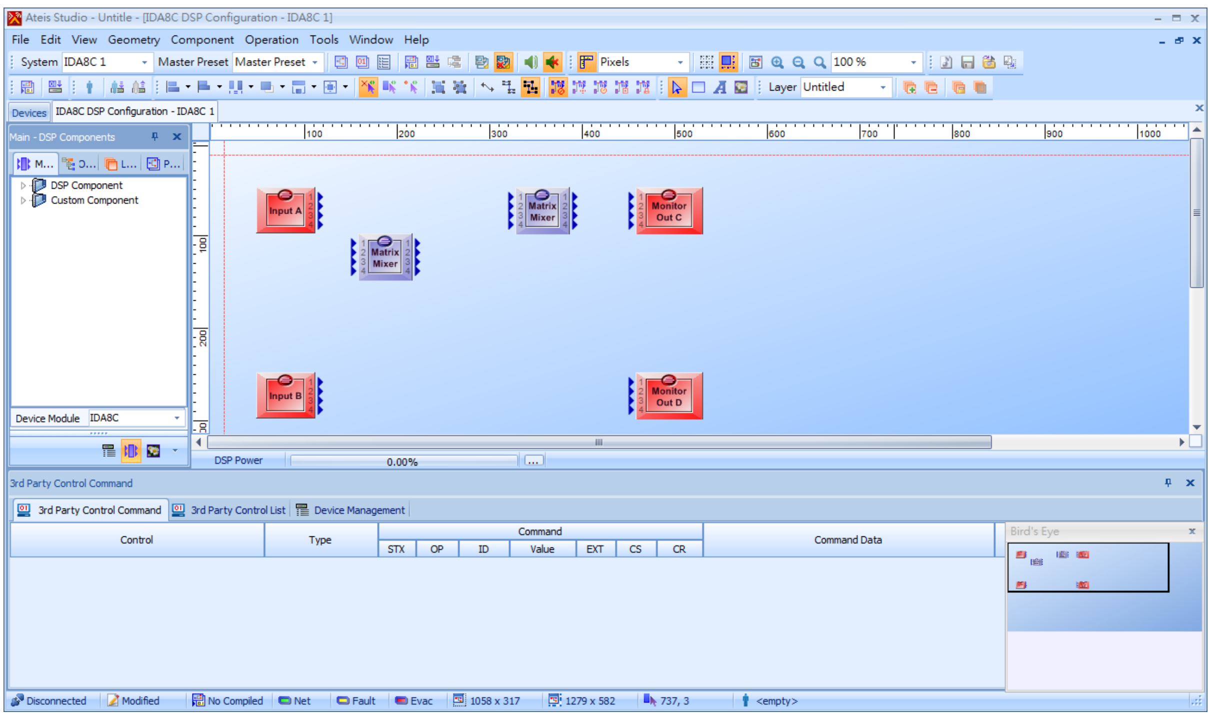
Close all the opened windows.

❖ UI Preset

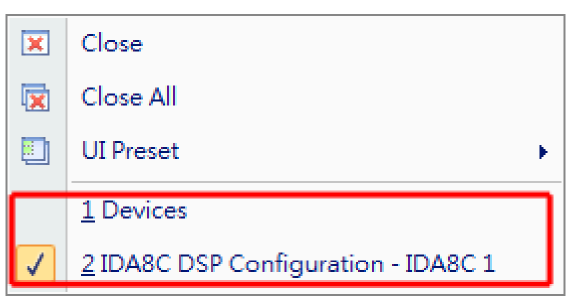
The UI Presets are pre-defined windows positioning.

- I'm Lost

By default the UI Preset "I'm Lost" is available, it allows coming back in the default standard windows positioning.



❖ Active Windows



In the bottom of the window menu, you will find all the opened windows of your project. By selecting of the windows in the list, the window will comes on the first displaying level.

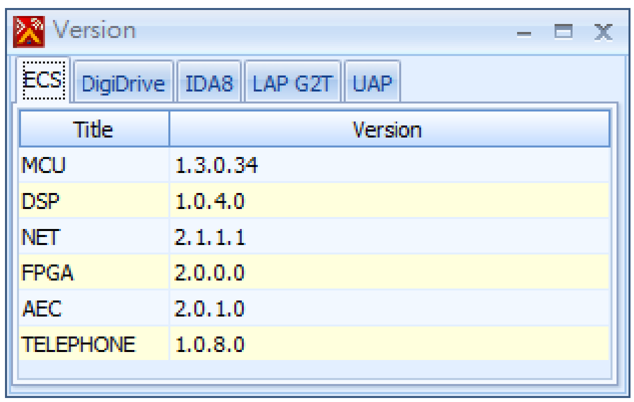
5.1.13 Help

❖ About

Open the About window.

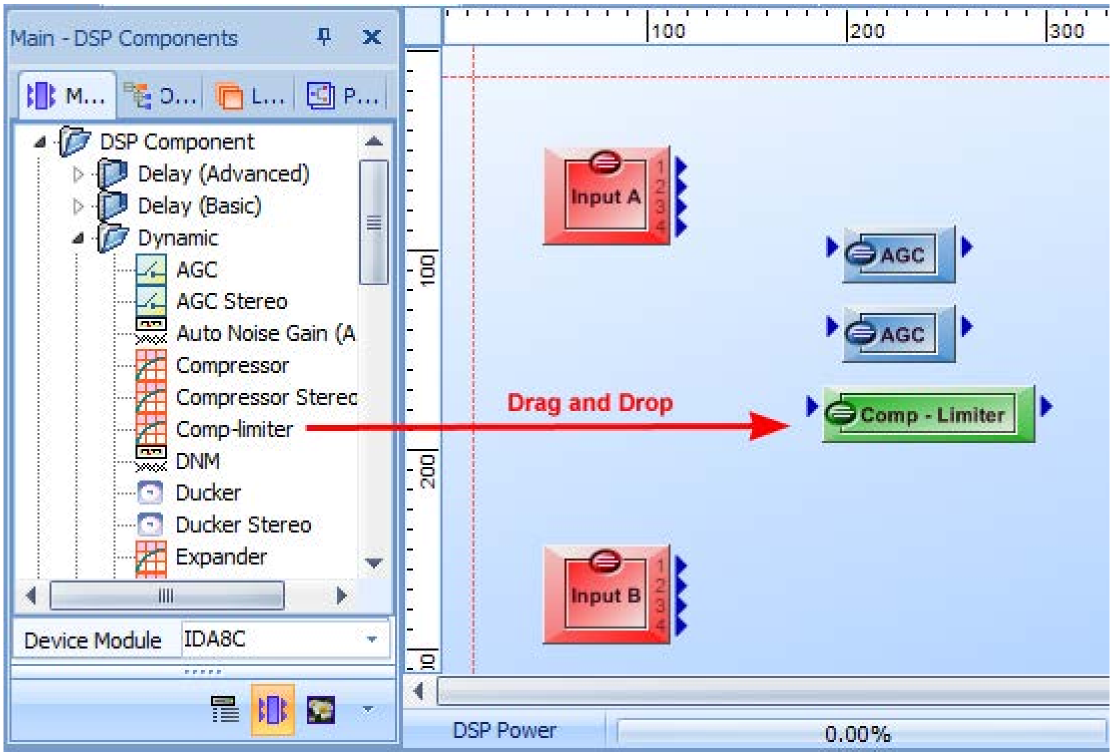
❖ Version

Open the Version window. This show compatible firmware version of devices that Ateis Studio support.



5.1.14 Component Template

In the configuration file of Ateis Studio, you can create a lot of objects to make up an system for the application. Each object works different jobs and you need to create it in design window to make it working. Component Template is a container for objects icons that ready to drag and drop down to design windows for creating an object for configuration. Below figure shows how component template work with design window:



On the left is component template and right is design window. There are three pages of component template, you can click icons on the bottom of component template to switch them.

❖ Devices Page

List devices which are ready for drag and drop to device design window. Some of object is corresponds to physical devices and some are just symbols.

❖ DSP Components Page

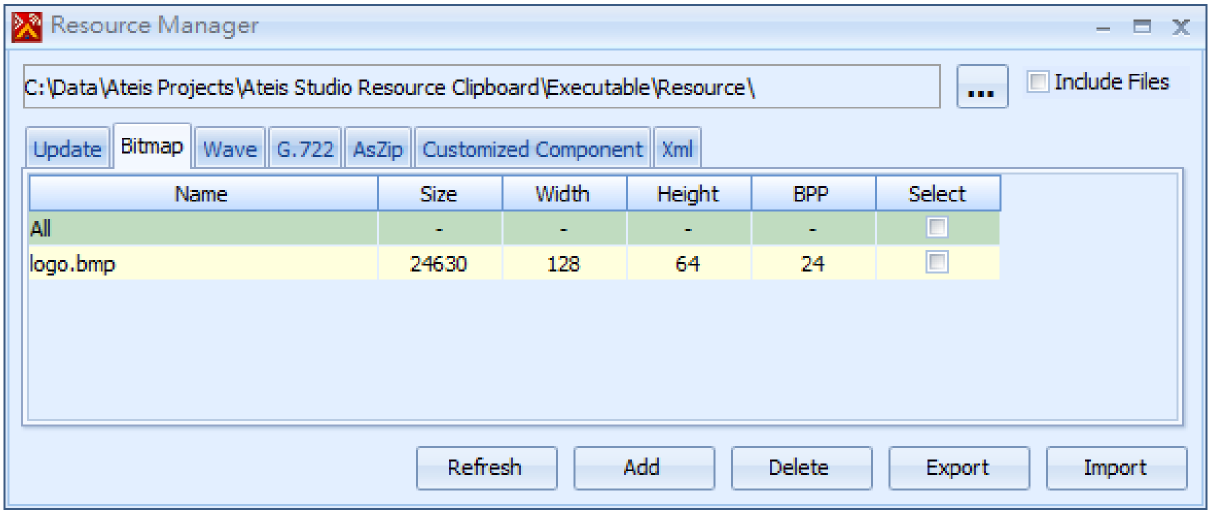
List DSP components which are ready for drag and drop to DSP design window. Each of them is corresponds to a signal processing unit inside the device.

❖ Extended Components Page

List symbolic components likes images, texts. All of them does not relates to physical functions.

By click menu [View > Main], you can open component template window.

5.1.15 Resource Manager



Resource manager is a container for files can be used in device configuration. Sections listed below describe each part of the resource manager settings window:

❖ Path for Resource Manager

Specify path for searching resource files.

❖ Browse the folder.

The ellipsis button to change path.

❖ Include file

When path changing, copy files from original path to new path.

❖ Pages of Resource File

There are various types of file can be managed by Ateis Studio, each of them corresponds to a page in resource manager. The files are listed below:

- Update
- Bitmap
- Wave
- G.722
- AsZip
- Customized Component
- xml

❖ Refresh

Search resource files in the folder specified in resource manager path.

❖ Add

Add a file.to resource manager.

❖ Delete

Remove selected files from resource manager.

❖ Export

Export selected files to a single resource packed file.

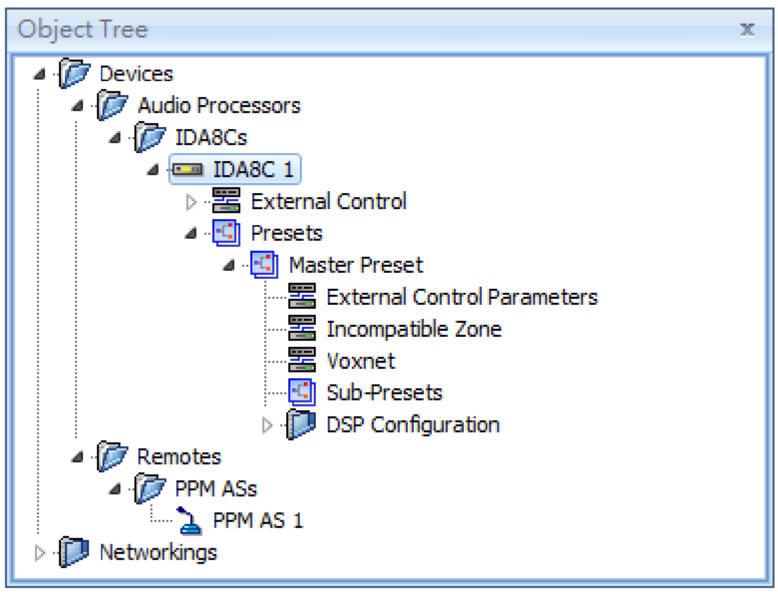
❖ Import

Import files from a resource packed file.

By click menu [View > Resource Manager], you can open resource manager window.

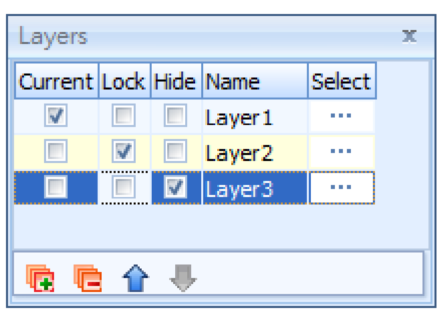
5.1.16 Object Tree

Object tree display designing objects by tree structure, make easy understanding and access objects. By click menu [View > Object Tree], you can open object tree window.



5.1.17 Layers

For each design windows, can be separate to multiple layers. every objects in design window belongs to a layer. By click menu [View > Layers], you can open layers window.



On top of windows, there is a grid to list all layers. The field are describes below:

❖ Current

Indicate which layer is current layer. The newly created objects will put into current layer.

❖ Lock

Lock selected layer. All objects belongs to the locked layer can't be edited.

❖ Hide

To hide all objects belongs to selected layer.

❖ Name

The name of layer.

❖ Select

An ellipsis button to select objects of the layer.

There are four buttons on bottom of window:

❖ Add Layer

Add a layer. The new layer becomes current layer.

❖ Remove Layer

Remove selected layer, and components belongs to the layer are removed too.

❖ Move up Layer

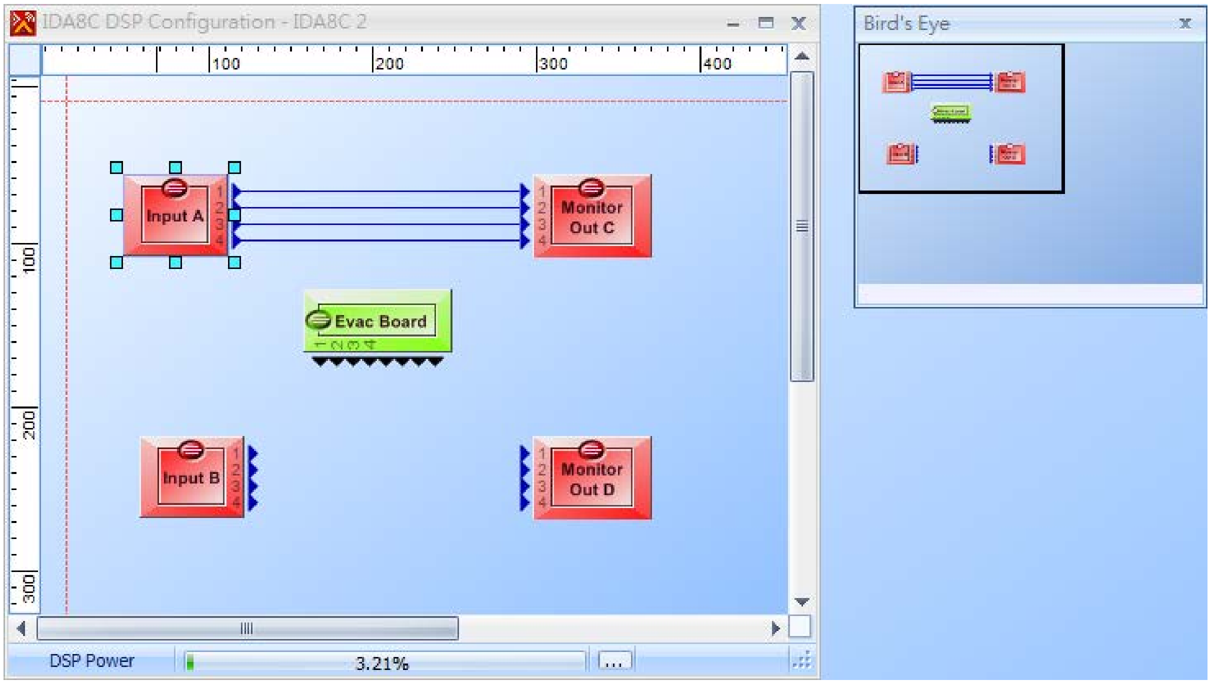
Move up selected layer a row.

❖ Move down layer

Move down selected layer a row.

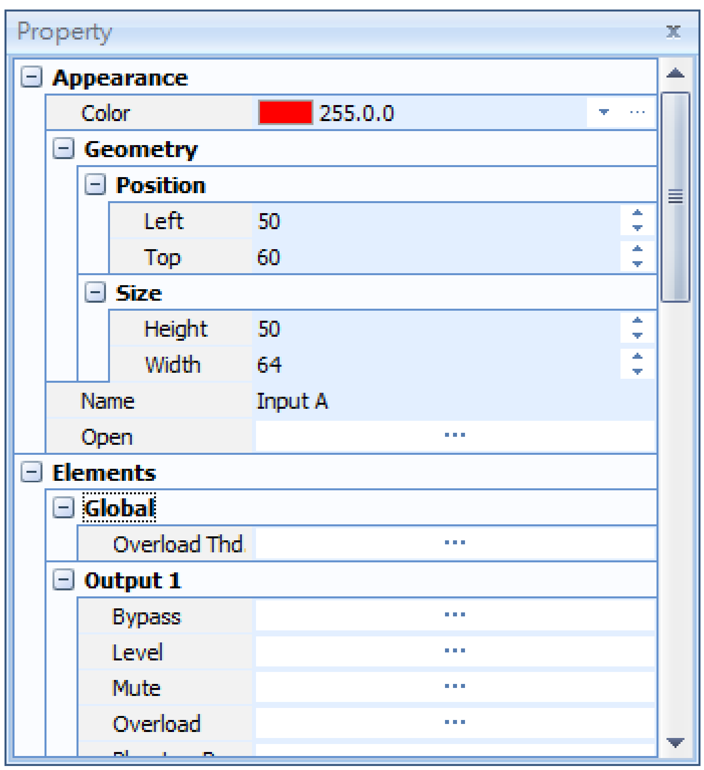
5.1.18 Bird's Eye

When the design area is huge, bird's eye gives a overview for user to easy understand the layout of design. By click menu [View > Bird's Eye], you can open bird's eye window.



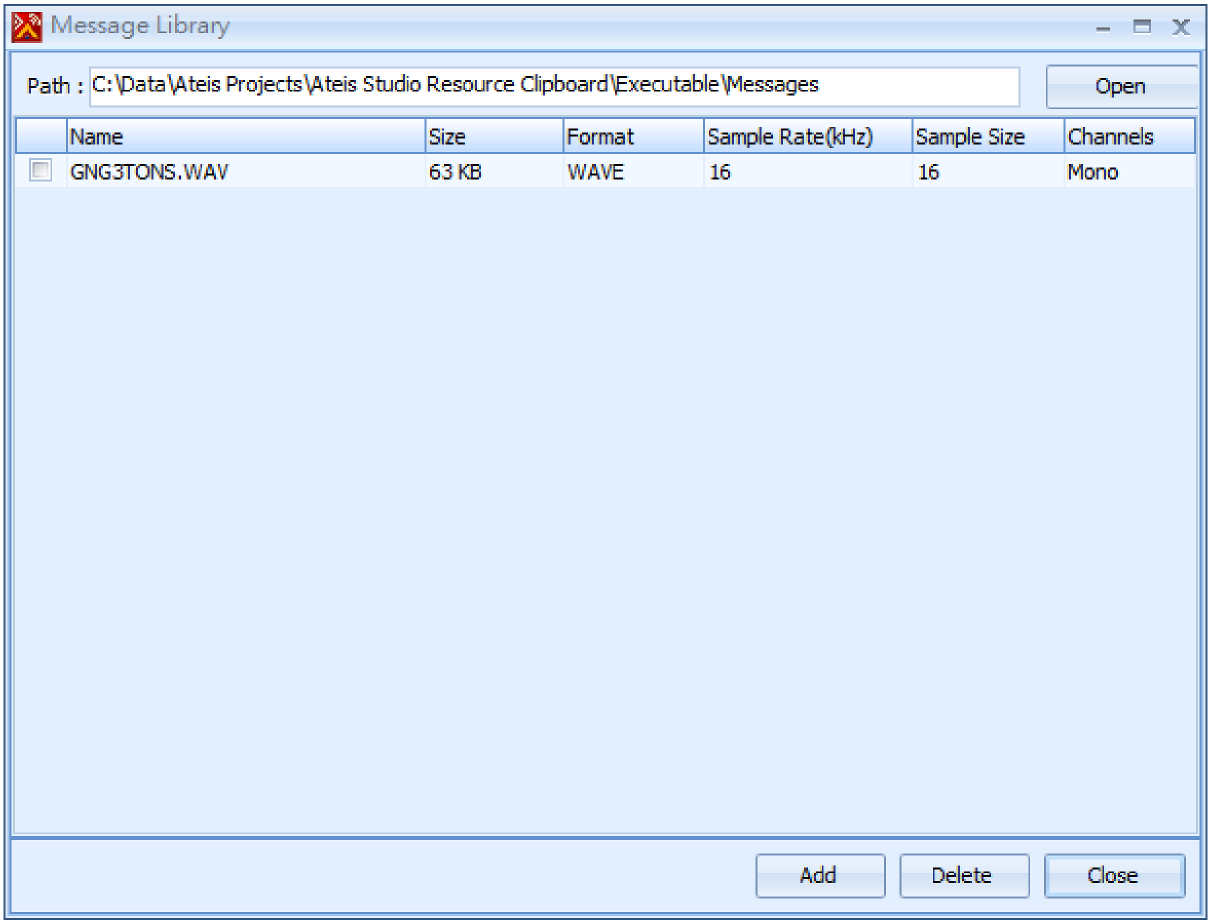
5.1.19 Properties

Properties window allow user to browse/edit settings of selected objects in design window. By click menu [View > Properties], you can open properties window.



5.1.20 Message Library

Message Library provides a easy interface to manage messages used in Ateis Studio. By click menu [View > Message Library], you can open message library window.



- Path

  The working directory for message files.

- Open

  Open a file browser to change working directory.

- Message File LIst

  A grid to list all messages in working directory, and display information about it.

- Add

  To add a message into working directory.

- Delete

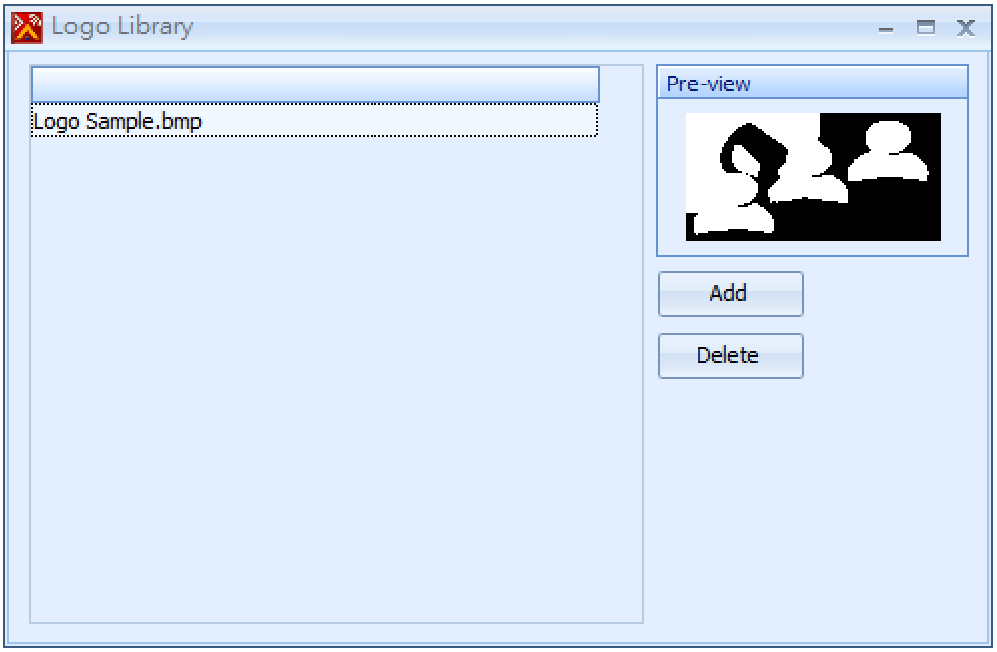
  Remove selected entry of message list.

- Close

  Close Message Libraray window.

5.1.21 Logo Library

Logo Library is a settings windows about Logo Library management. By click menu [View > Logo Library], you can open logo library window.



The left of window list all Logo files.

On right of window there is a preview window for selected logo file. Button [Add] to add a logo file, Button [Delete] to remove selected logo file.

5.2 Presets

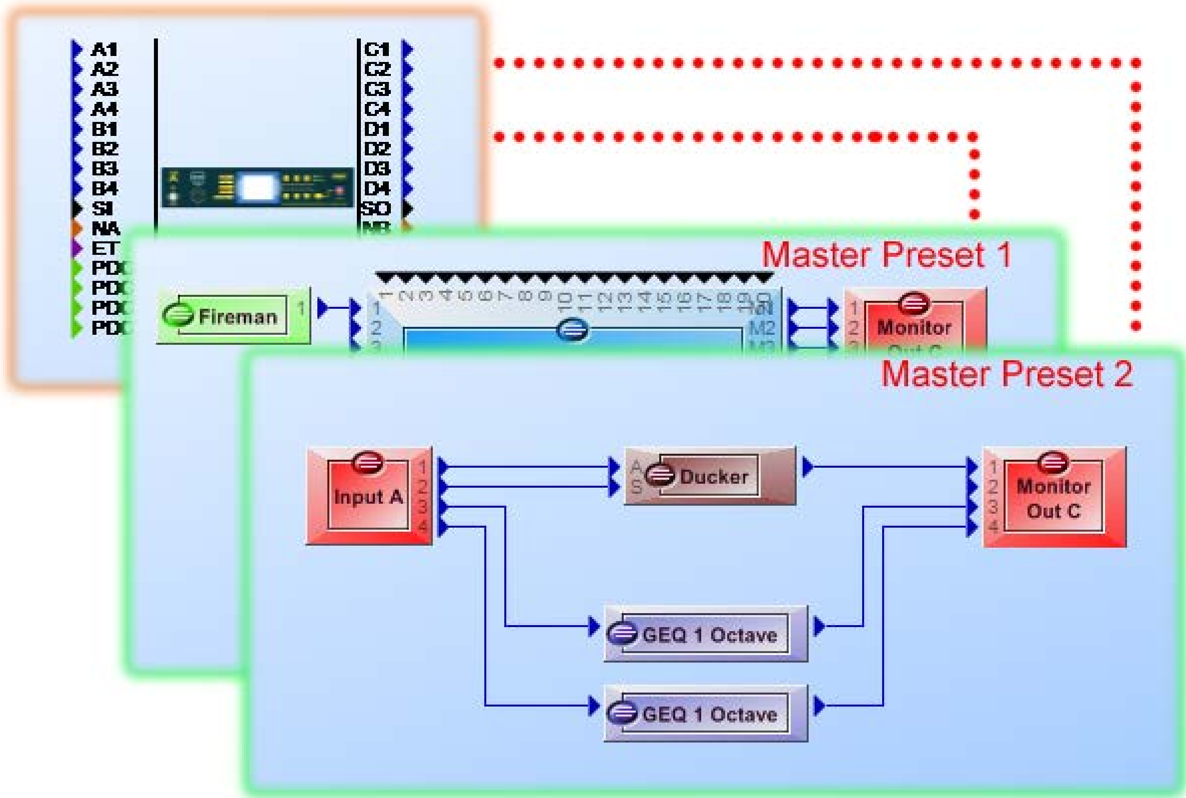
5.2.1 Overview

The Ateis systems include two types of presets:

- The Master presets (design preset): they enable completely different designs to be restored.
- The Sub-Presets (parameters preset): They enable values of multiple parameters of the same design, such as levels, gains, EQ, etc. to be restored either from the PC software, the remote controllers or the control inputs.

These 'Master Presets' and 'Sub-Presets' residing in the devices memory which can be accessed in many ways: PC, Logic inputs, Third party. The max number of master presets and sub presets depends on machine memory size and content of each preset.

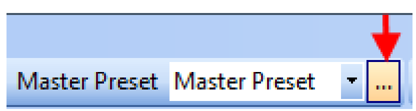
5.2.2 Master Presets



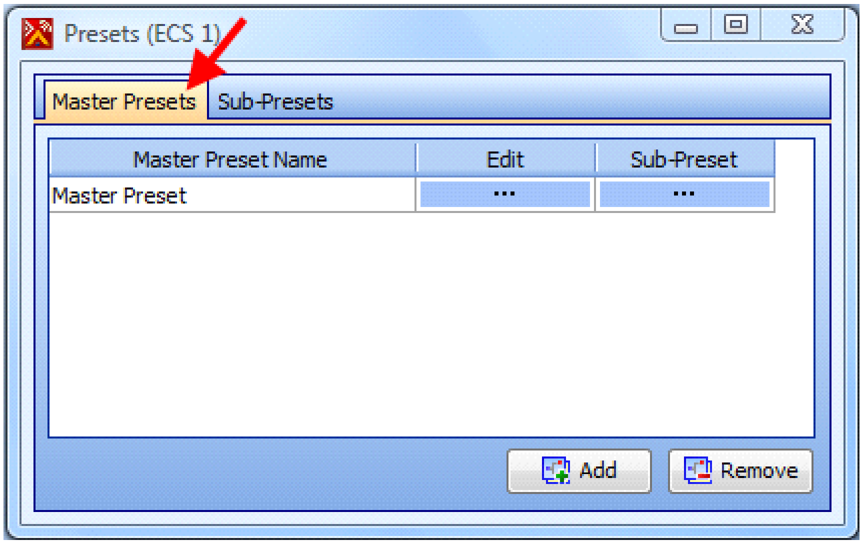
The Master-Preset function allows switching between several designs on the same device. Each Master-Preset can contains different components, design path, sub-preset, etc. Then the user can switch of the complete design simply by change of Master-Presets. That can be done by using logic input contact or by third party.

❖ Master Preset Management

This section describe how to create or remove master presets. On the main window top, click on the ellipsis button next to Master Preset combo box.



The Presets window opens. Select the "Master-Presets" tab.

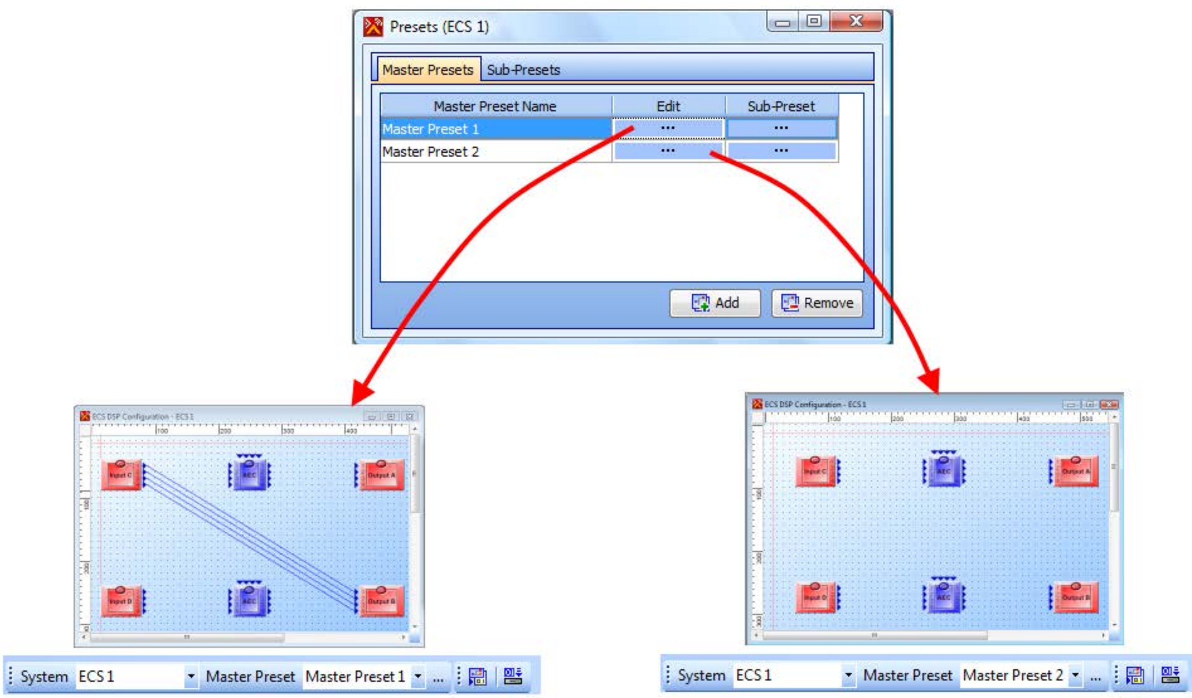


- Master Preset Name

  The name of master preset.

- Edit

  By clicking ellipsis cell in grid you can open the dsp design window of the preset.



- Sub-Preset

  Open the settings window of sub-preset which is belong to the master preset.

- Add

  To create a new master preset.

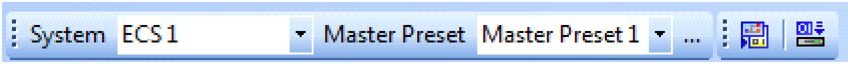
- Remove

  To remove selected master preset.

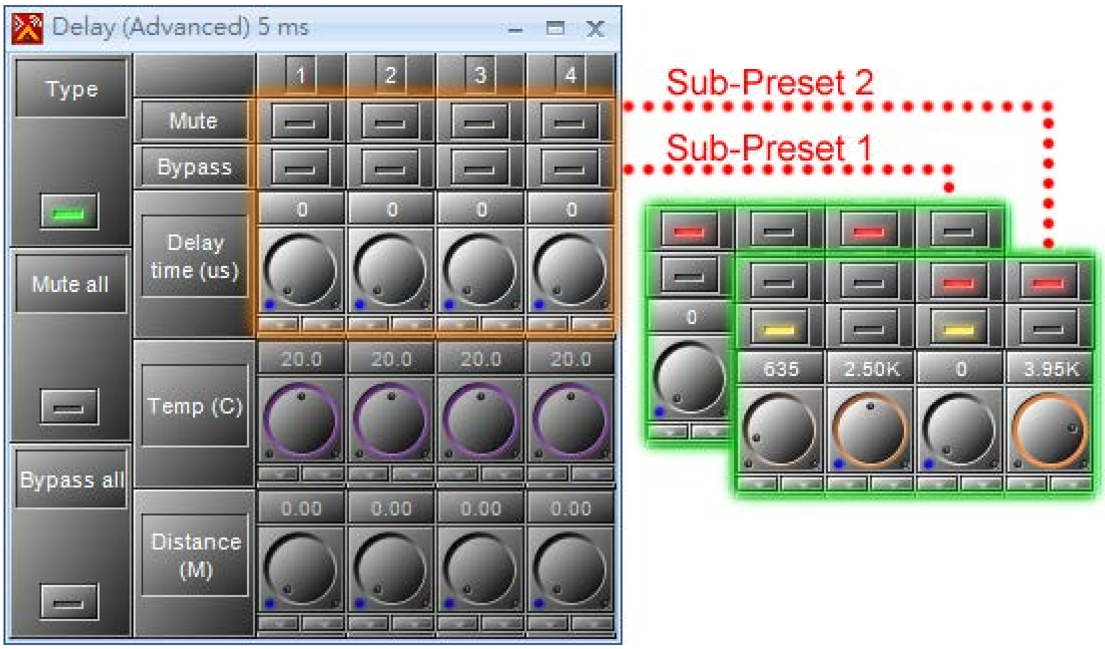
❖ Master Preset Switching

There is always only one master preset active at the same time. When you're designing a system, change master to the preset you want to edit. At run time, change master preset to make Ateis device working on different DSP design belong to the master preset.

Click combo box Master Preset to switch master preset:



5.2.3 Sub-Presets

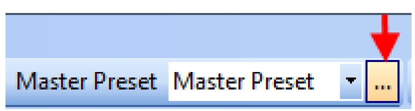


The sub-Preset function allows storing values of different component elements (volume gain, frequency, mute, etc) in one Sub-Preset. By loading a Sub-Preset, all the elements associated will take the predefined values. The Sub-Presets are different in each Master-Presets and of course in each System (Device).

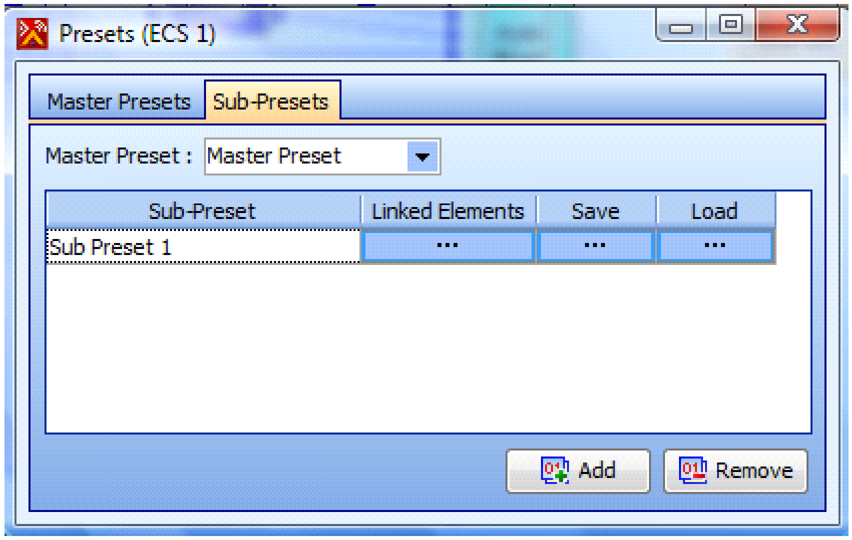
The idea is to create a Sub-Preset, define which element of which component you want to associate with the Sub-Preset, then to set this element to their target values, and then to memorize that in the Sup-Preset.

❖ Sub-Preset Management

On the main window top, click on the ellipsis button next to [Master Preset] combo box.



The Presets window opens. Select the "Sub-Presets" tab.



- Master Preset Combo Box

  By clicking this combo box, you can switch to desired master preset and then edit sub-preset settings belonging to it.

- Sub-Preset

  Name of the sub-preset.

- Linked Elements

  Open the window containing the table of elements controlled by sub-preset. see later section "Table of linked elements" for more detail.

- Save

  Save value of linked elements to sub-preset. Those values are kept in the memory storage in the sub-preset.

- Load

  Load values from the storage of sub-preset and set to linked elements.

- Add

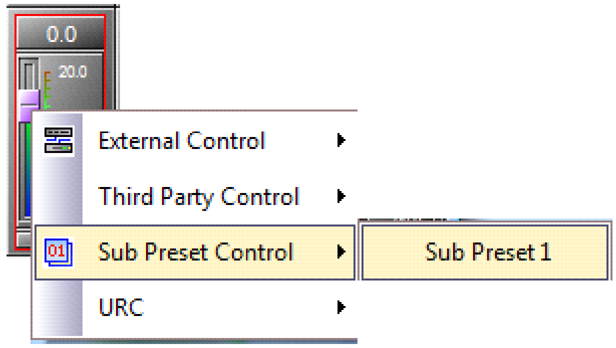
  Add a sub-preset.

- Remove

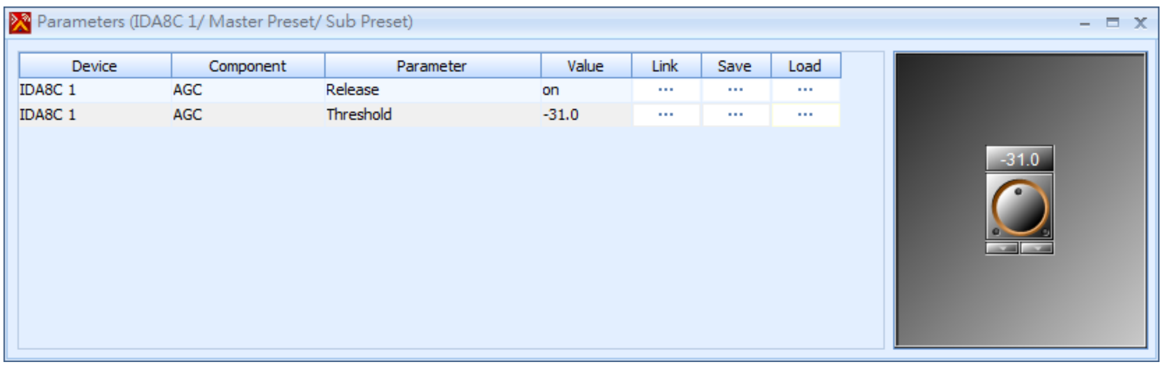
  Remove a sub-preset.

❖ Assign an element to be controlled by sub-presets

Click right on the element of the component you want to control, and select your sub-preset.



❖Table of linked elements



This table list all elements controlled by the sub-preset, on the right there is a small panel embedded a graphic control for adjusting the stored value in sub-preset. Each field is described in below sections.

- Device

  Indicate the device containing the linked element.

- Component

  Indicate the component containing the linked element.

- Parameter

  The element be controlled by the sub-preset.

- Value

  Value stored in sub-preset.

- Link

  To open the window containing the linked element.

- Save

  To save current value of elements to the sub-preset.

- Load

  To load value stored in sub-preset and set to linked elements.

5.3 Event Management

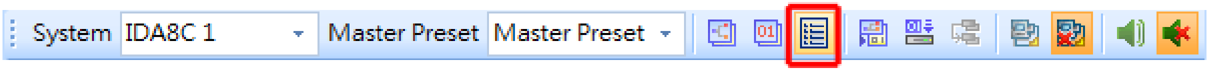
5.3.1 Event Manager

There are various type of events for controlling Ateis devices. For example, to adjust value of elements or change presets. Each event is able to link with a controller, The following list are sources able to link events:

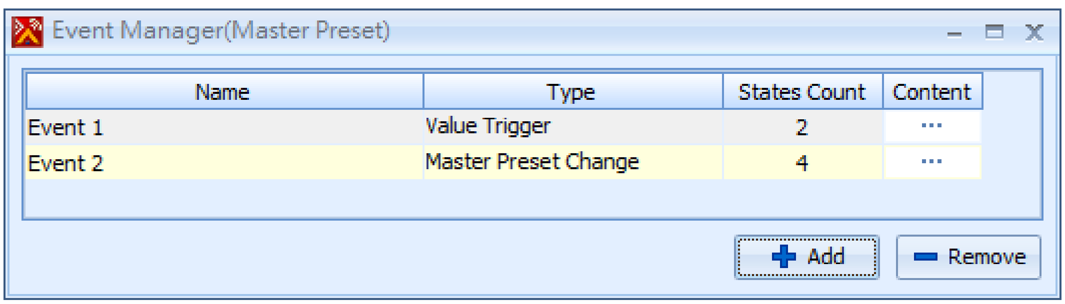
- Event Trigger component
- Event Value Control component
- PPM AS
- PSS AS
- Terra Controller
- Front Panel on some Ateis devices, like IDA8C

Event manager is an container to store all events and manage them. There is one event manager for each master preset.

Open Event manager by click  on tools bar:



The settings window shows below:



- Name

  The name of an event.

- Type

  The type of an event, different type of event executes different tasks when it triggers.

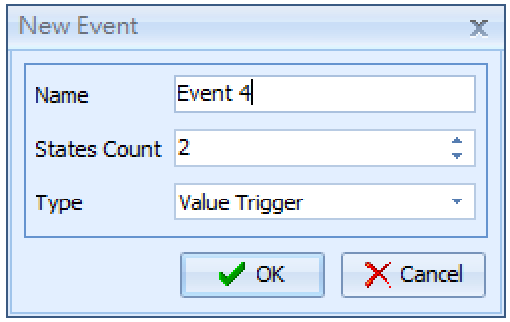
- States Count

  Some of events trigger with a state number,. For example, to control a element like mute you need to set State Count to 2, because mute have two states On/Off. States count is the number of states.

- Content

  Open settings window belong the event.

- Add



Add an event to event manager, Some event can't create in this window, like paging event, you need to create it in settings windows of Network Paging component.

- Remove

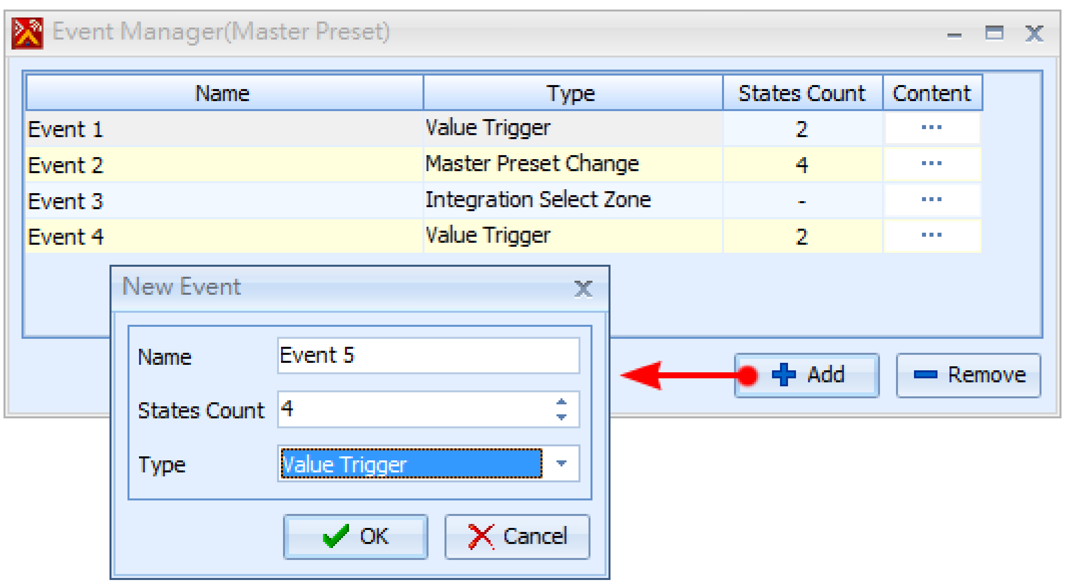
Remove selected event.

5.3.2 Value Trigger Event

❖Description

This event is used to control the element value. It set value of element by state number send by source that triggers the event. An value trigger event can link more than one elements for controlling.

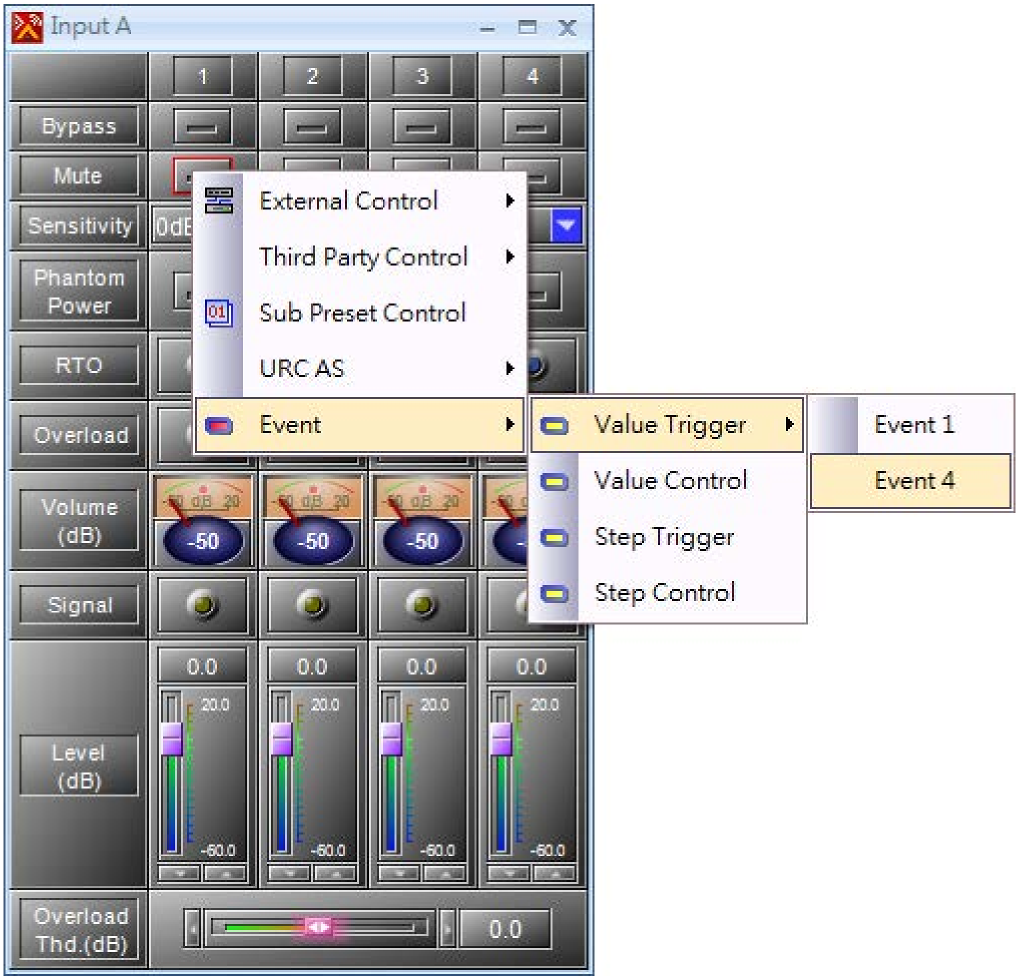
❖Event Creation



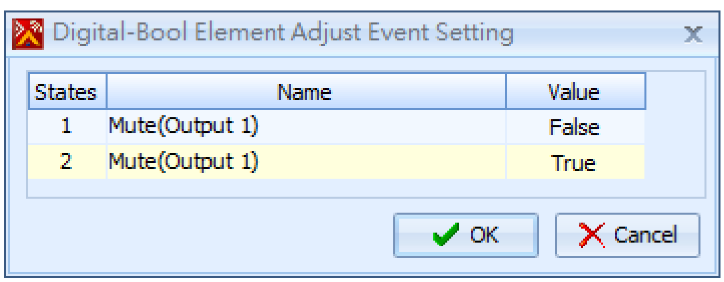
Click button [Add] to create a new event, a second window open to query you which kind of event is desired. Select item [Value Trigger] on Type combo box. Then, press button [OK].

❖Event Settings

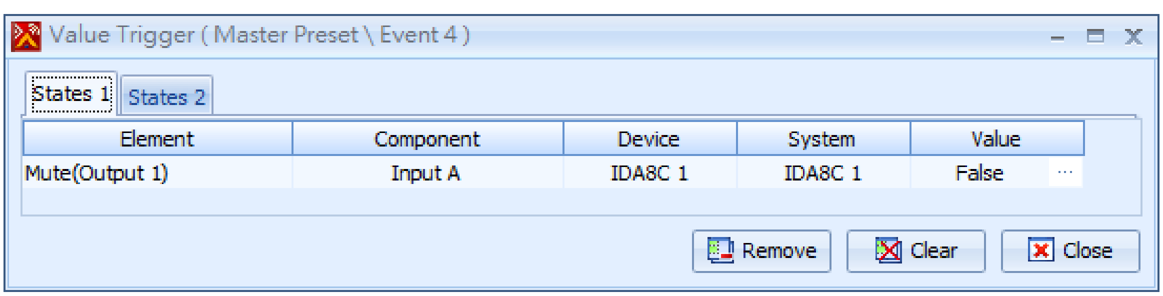
After create a value trigger event, you need to assign an element to it. Then the assigned element will be controllable by the event. To do assignment, right click on the desired element, click popup menu by sequence: [Event > Value Trigger > XXX] where XXX is the name of event. The following figure is an example.



Then a window shows brief information of the relationship state and value:



Click field [Content] of event manager window to open settings window of the event:



- State Pages

  There is a page for each state in value trigger event. To switch states, click the tab of desired state.

- Element

  Indicate which element of the component controlled by value trigger.

- Component

  Indicate which component of the device contains element prior mentioned.

- Device

  Indicate which device contains component prior mentioned.

- System

  Indicate which system contains device prior mentioned.

- Value

  Specify the value bind with the state. When a Value Trigger Event triggers, the source that triggers event send a state number to event system, event map this state number to a value and then set it to element.

- Remove

  Remove the element controlled by the event.

  ⚠ This operation only remove the relation between element and event, not delete element.

- Clear

  Remove all elements controlled by the event.

- Close

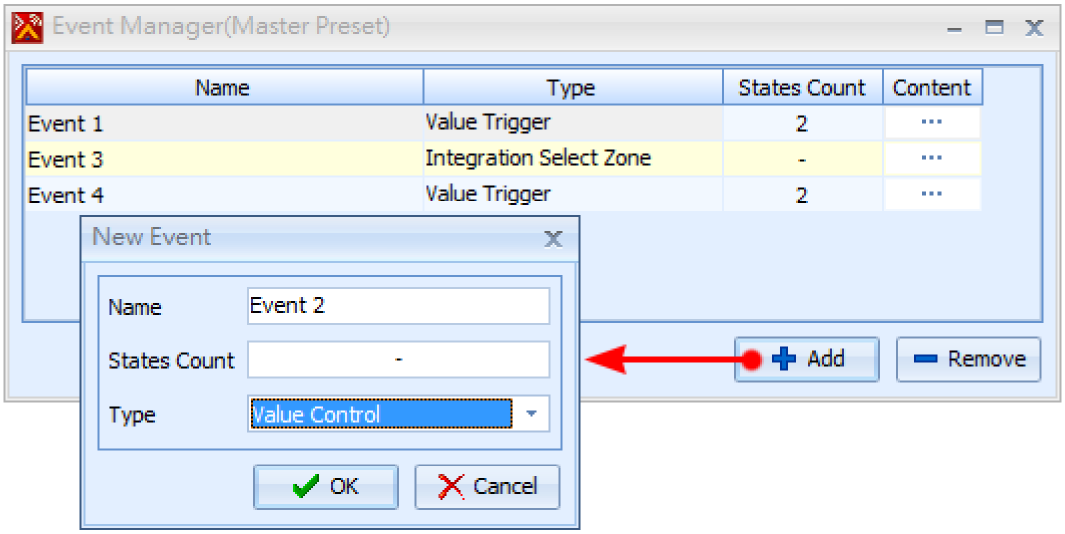
  Close the settings window.

5.3.3 Value Control Event

❖ Description

This event is used to control the element value. It set value of element by source that triggers the event. An value control event can link more than one elements for controlling.

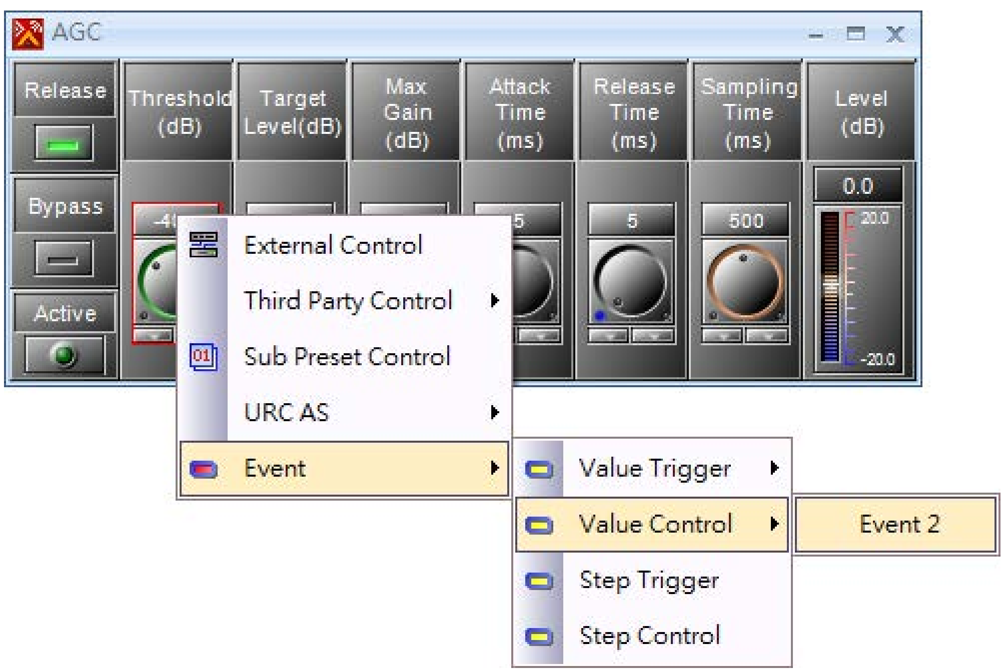
❖ Event Creation



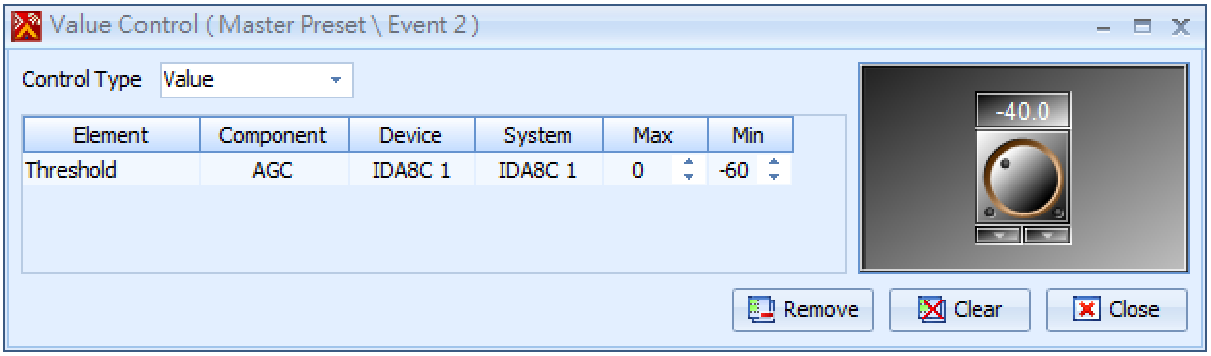
Click button [Add] to create a new event, a second window open to query you which kind of event is desired. Select item [Value Control] on Type combo box. Then, press button [OK].

❖ Event Settings

After create a value control event, you need to assign an element to it. Then the assigned element will be controllable by the event. To do assignment, right click on the desired element, click popup menu by sequence: [Event > Value Control > XXX] where XXX is the name of event. The following figure is an example.



Click field [Content] of event manager window to open settings window of the event:



On the right side of window, there is a knob to simulate source that triggers event. If you adjust the knob, the value of elements also changed.

- Control Type

  This option determines the type of knob simulation, there are two options of Control Type:

  - Value

    Knob adjust value of elements using native value.

  - Percent

    Knob adjust value of elements by percentage.

- Element

  Indicate which element of the component controlled by value trigger.

- Component

  Indicate which component of the device contains element prior mentioned.

- Device

  Indicate which device contains component prior mentioned.

- System

  Indicate which system contains device prior mentioned.

- Min

  Specify the minimum value of simulating knob.

- Max

  Specify the maximum value of simulating knob.

- Remove

  Remove the element controlled by the event.

  This operation only remove the relation between element and event, not delete element.

- Clear

  Remove all elements controlled by the event.

- Close

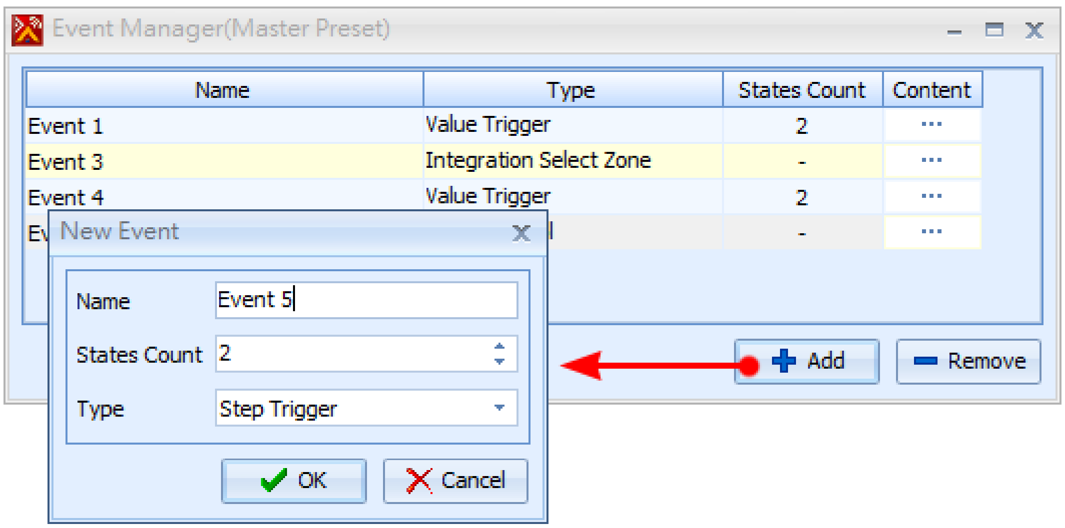
  Close the settings window.

5.3.4 Step Trigger Event

❖ Description

This event is used to control the element value. It adjust value of element by step when source triggers the event. There are multiple states in an step trigger event, each state contains a step value.

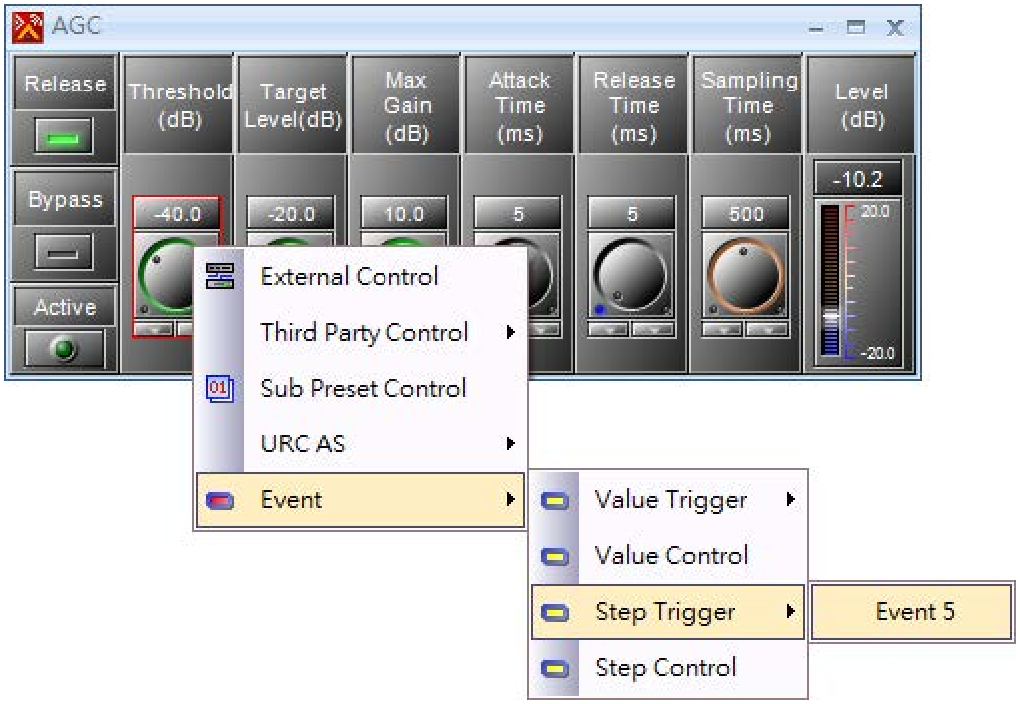
❖ Event Creation



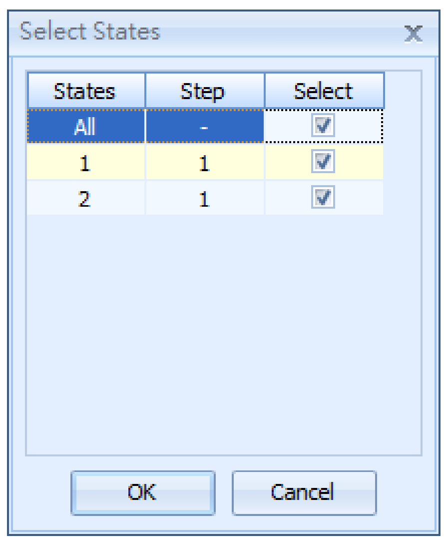
Click button [Add] to create a new event, a second window open to query you which kind of event is desired. Select item [Step Trigger] on Type combo box. Then, press button [OK].

❖ Event Settings

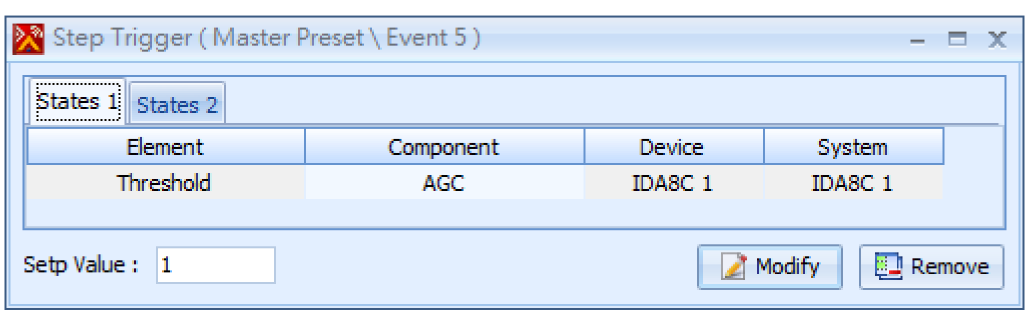
After create a value control event, you need to assign an element to it. Then the assigned element will be controllable by the event. To do assignment, right click on the desired element, click popup menu by sequence: [Event > Step Trigger > XXX] where XXX is the name of event. The following figure is an example.



Then a window shows brief information, click select check box at row "All":



Click field [Content] of event manager window to open settings window of the event:



- State Pages

  There is a page for each state in step trigger event. To switch states, click the tab of desired state.

- Element

  Indicate which element of the component controlled by value trigger.

- Component

  Indicate which component of the device contains element prior mentioned.

- Device

  Indicate which device contains component prior mentioned.

- System

  Indicate which system contains device prior mentioned.

- Step Value

  Specify step value of the state.

- Modify

  Enable or disable selected element controlled by state of event.

- Remove

  Remove the element controlled by the event.

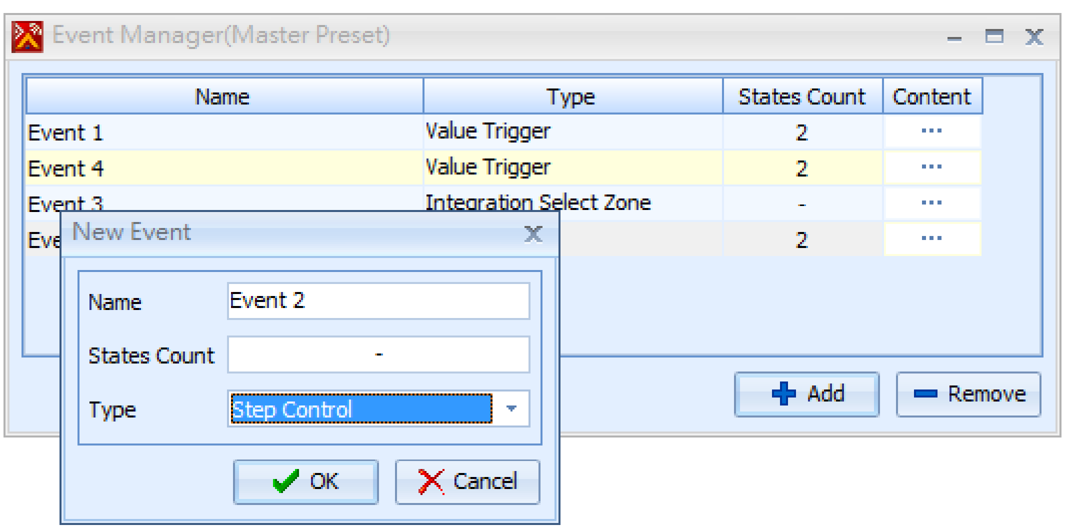
  ⚠ This operation only remove the relation between element and event, not delete element.

5.3.5 Step Control Event

❖ Description

This event is used to control the element value. It adjust value of element by step when source triggers the event. The source triggers event send the step value to event for adjusting elements.

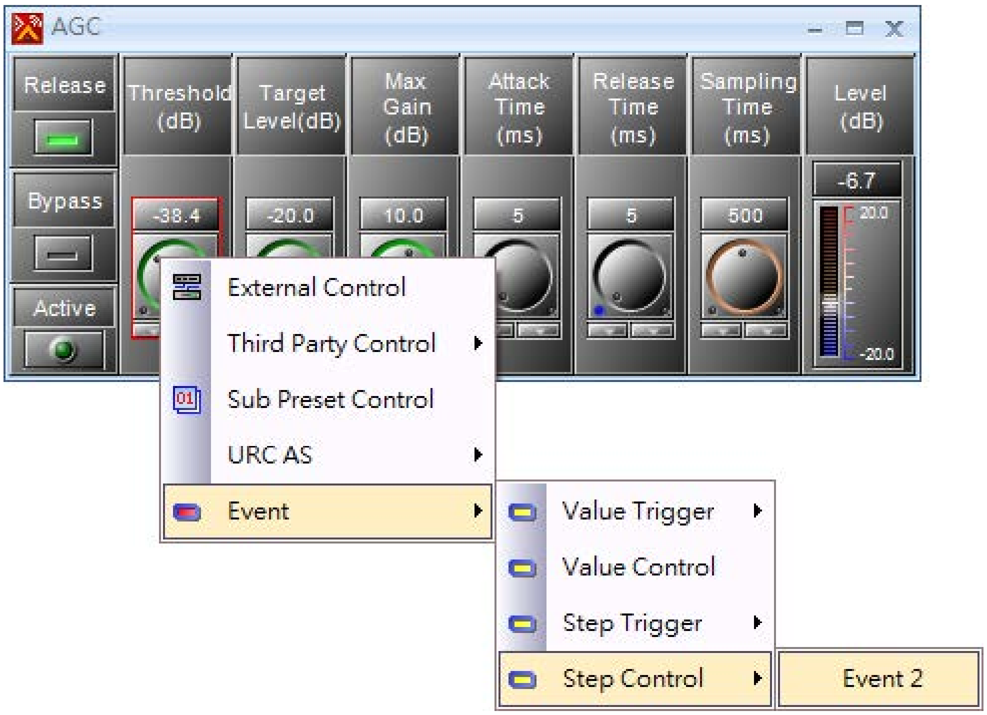
❖ Event Creation



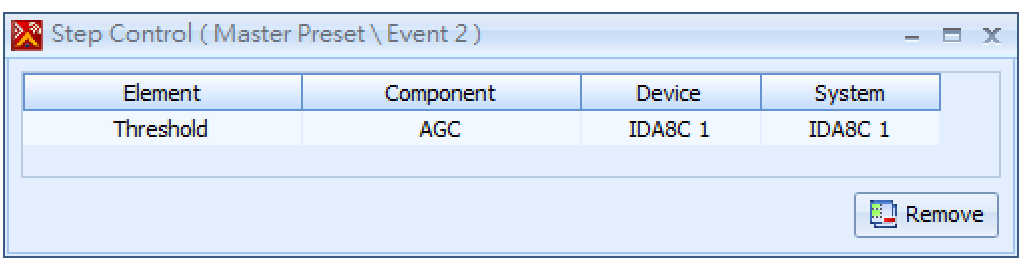
Click button [Add] to create a new event, a second window open to query you which kind of event is desired. Select item [Step Control] on Type combo box. Then, press button [OK].

❖ Event Settings

After create a value control event, you need to assign an element to it. Then the assigned element will be controllable by the event. To do assignment, right click on the desired element, click popup menu by sequence: [Event > Step Control > XXX] where XXX is the name of event. The following figure is an example.



Click field [Content] of event manager window to open settings window of the event:



- Element

  Indicate which element of the component controlled by value trigger.

- Component

  Indicate which component of the device contains element prior mentioned.

- Device

  Indicate which device contains component prior mentioned.

- System

Indicate which system contains device prior mentioned.

- Remove

  Remove the element controlled by the event.

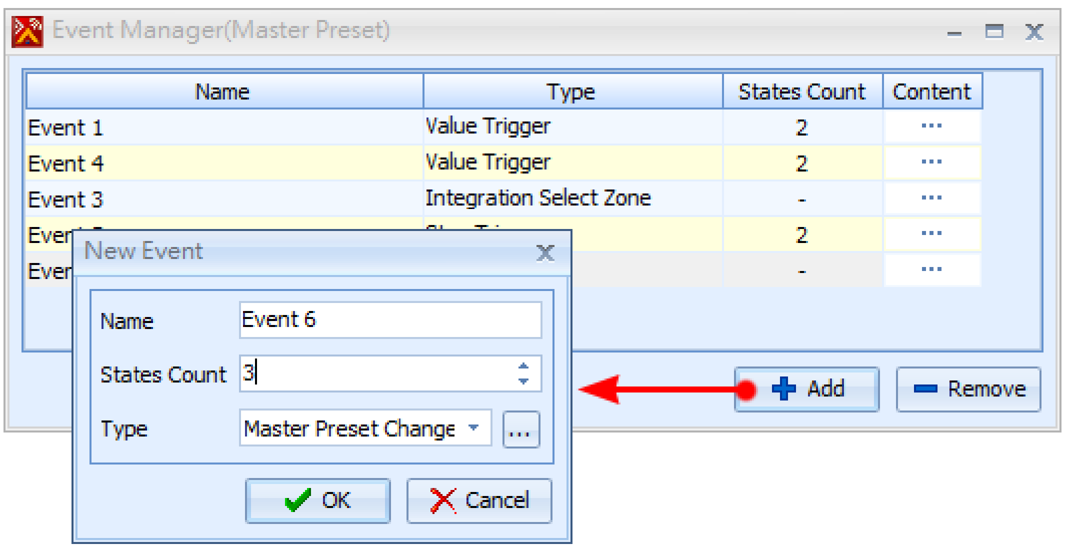
  ⚠ This operation only remove the relation between element and event, not delete element.

5.3.6 Master Preset Change Change

❖ Description

This event is used to change master preset of Ateis devices. Each state can be linked to a master preset. When a source triggers a master preset change event, it send state number to event, then change to the master preset linked to the state.

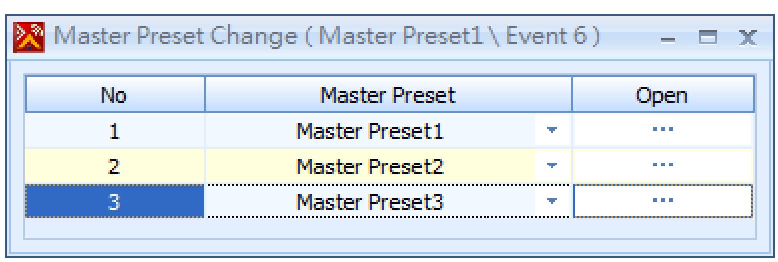
❖ Event Creation



Click button [Add] to create a new event, a second window open to query you which kind of event is desired. Select item [Master Preset Change] on Type combo box. Then, press button [OK].

❖ Event Settings

Click field [Content] of event manager window to open settings window of the event:



- No

  Indicate state No.

- Master Preset

Specify which master preset links to the state.

- Open

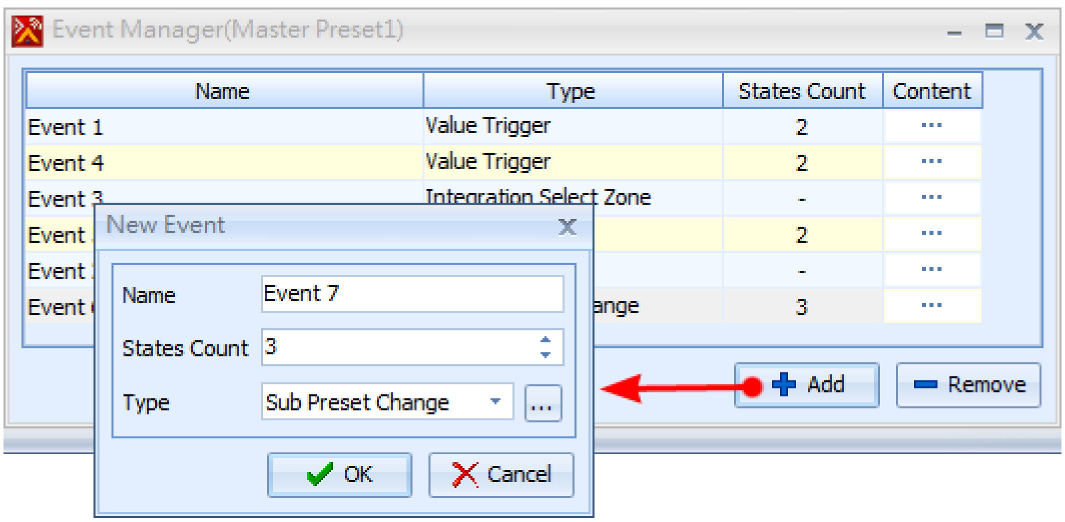
  Open settings window of the master preset.

5.3.7 Sub-Preset Change Evnet

❖Description

This event is used to change sub-preset of Ateis devices. Each state can be linked to a sub-preset. When a source triggers a sub-preset change event, it send state number to event, then change to the sub-preset linked to the state.

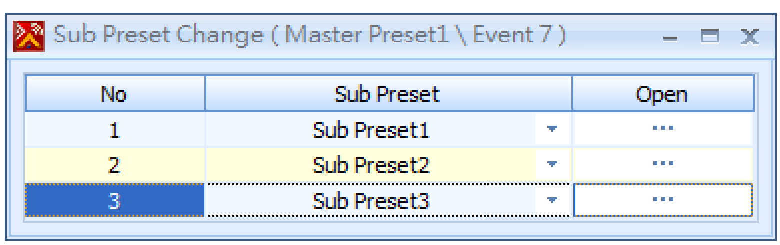
❖Event Creation



Click button [Add] to create a new event, a second window open to query you which kind of event is desired. Select item [Sub Preset Change] on Type combo box. Then, press button [OK].

❖Event Settings

Click field [Content] of event manager window to open settings window of the event:



- No

  Indicate state No.

- Sub Preset

  Specify which sub-preset links to the state.

- Open

Open settings window of the sub preset.

5.3.8 Intergration Paging Event

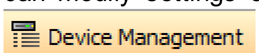
This event is used to select zones of Networking Paging component with multiple combination before paging. see the topic Network Paging for more details.

5.3.9 Singular Paging Event

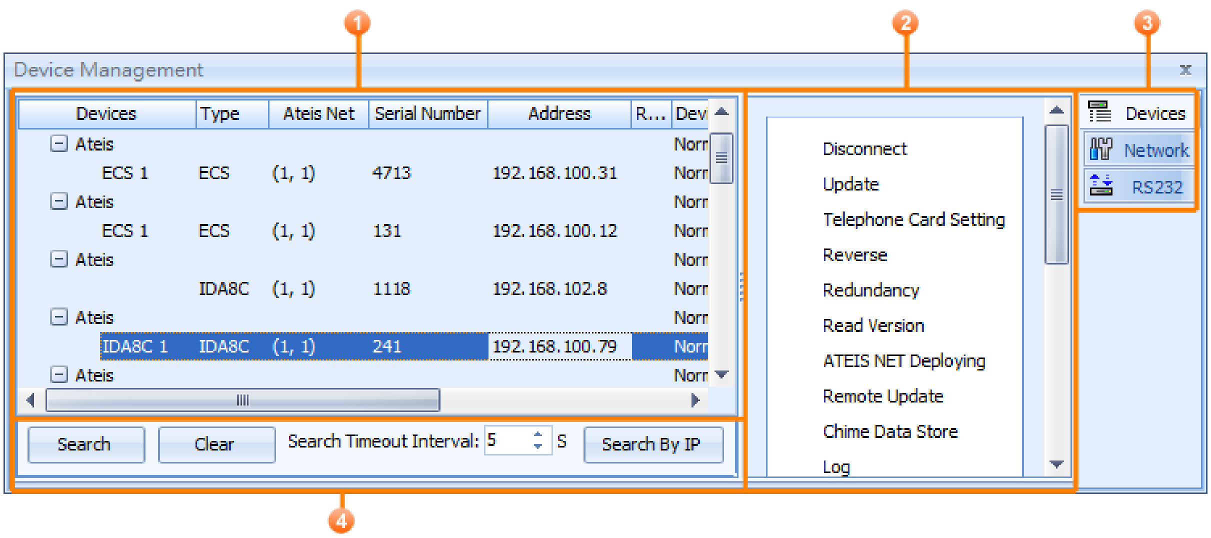
This event is used to select zones of Networking Paging component with single combination before paging. see the topic Network Paging for more details.

5.4 Device Management

5.4.1 Search and Settings

Device Management window help user to discover Ateis devices in the network, and shows information about it. You can modify settings of user by using the command on the window. Select the [Device Management] 

tab or open it through the menu [View > Device Management].



The Device Management window is consist of four part:

1. Device Information

List information of the device, including following fields:

- Devices

  Discovered devices are listed using tree structure.

- Type

  Type of device.

- Ateis Net

  Display Ateis Net ID by (X, Y) format, where X is the global net ID and Y is local net ID.

- Serial Number

  Serial Number of the device. Each Ateis device has an unique serial number.

- Address

  Shows IP address of the device.

- Redundancy Addr

  The redundancy address of the device.

- Device Status

  Show the status of devices. If there is a fault in audio processor, this field shows "Fault".

- Name

  The object of design window that corresponds to the device. After the relation of object and device created, the configuration of object will set to the device after execute "store" procedure.

- Communication

  Show the communication state of Ateis Studio and device.

2. Device Function

   List all functions available of the device.

3. Settings Tab

   Switch different tab for desired settings. There are thee tables :

   - Device

     Search and list Ateis devices in network.

   - Network

     Network interface card settings for communicating with Ateis devices.

   - RS232

     RS232 interface settings for communicating with Ateis devices.

4. Search Devices

   - Search

     Try to discovery Ateis devices on the network.

   - Clear

     Clear the current searched result, make the grid be blank.

   - Search Timeout Interval

     Determine how long should wait for device to reply for search command.

   - Search by IP

If you already known the IP address of a device, you can search it by that address.

Select the [Device Management] tab or open it through the menu [View > Device Management] to open Device Management window.

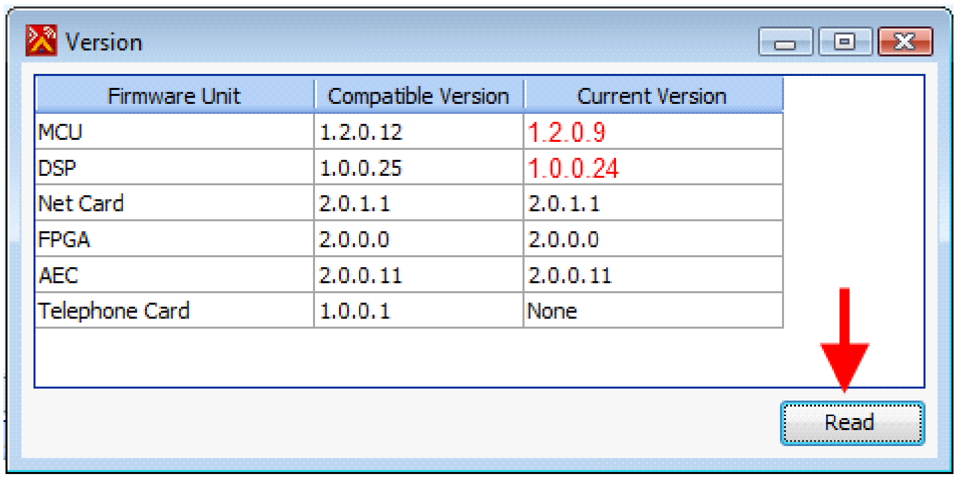
5.4.2 Read Version

To know which versions are loaded in the devices and which version are available with the current Ateïs Studio software:

1. Open Device Management Window.

2. Search and Connect to the device.

3. Click [Read Version] to open Version window.



Click [Read] button to get version data from the device.

- Firmware Unit

  The firmware program unit in the device.

- Compatible Version

  Displays the version required by the installed software.

- Current Version

  Displays the firmware version of the device.

5.4.3 Update Firmware

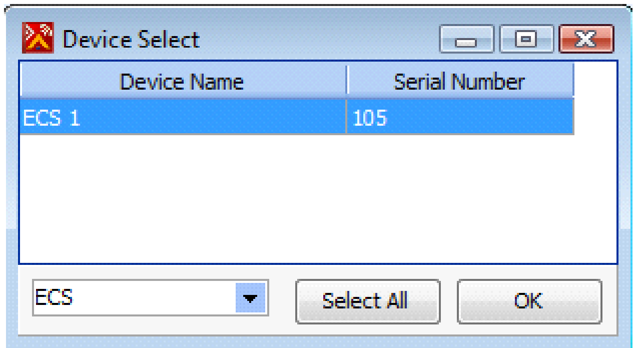
The purpose of update firmware is to match the version contained in your Ateïs Studio software with the version contained in the hardware of the devices.

Read version to know if there is versions mismatching. Then, if needed, proceed to the update. Updating the Ateïs Studio software sometimes makes necessary to update the audio processor firmware also. To do this, follow these steps:

1. Open Device Management Window.

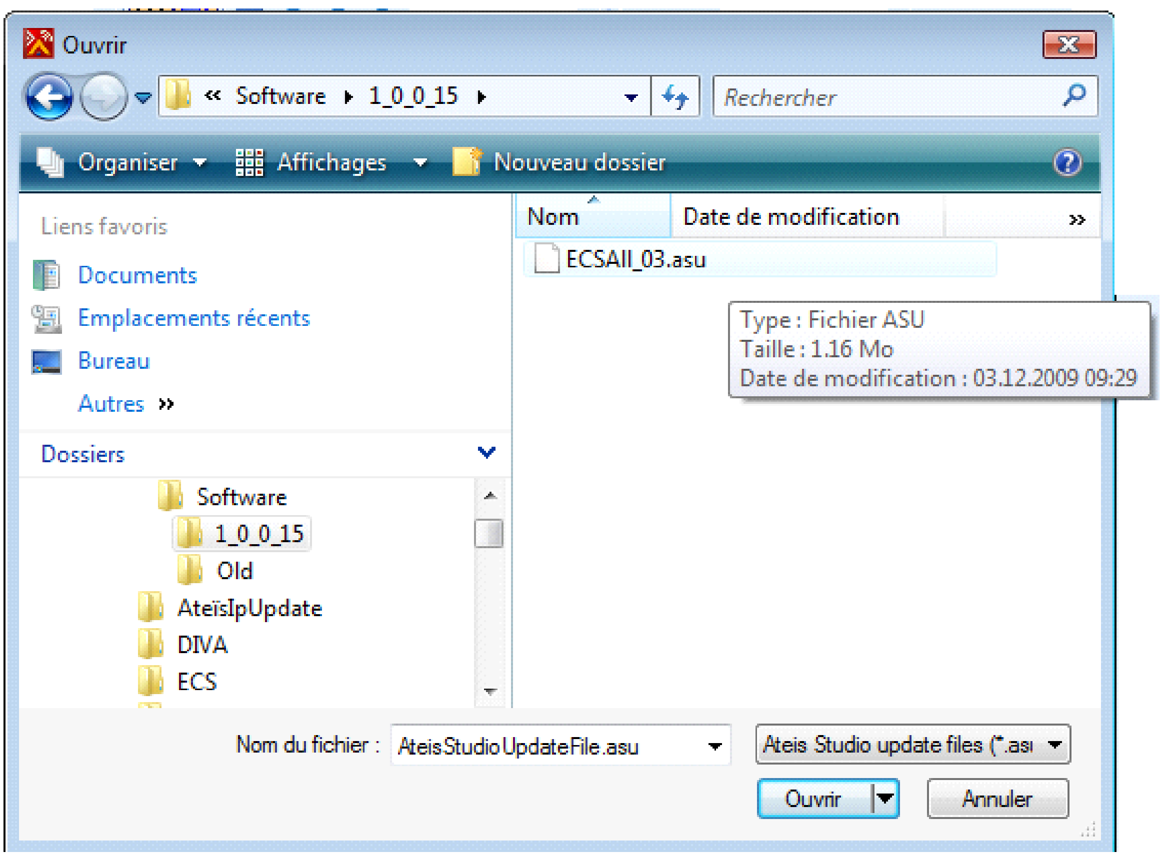
2. Search and Connect to the device needs to update.

3. Click [Update] to open the update window.

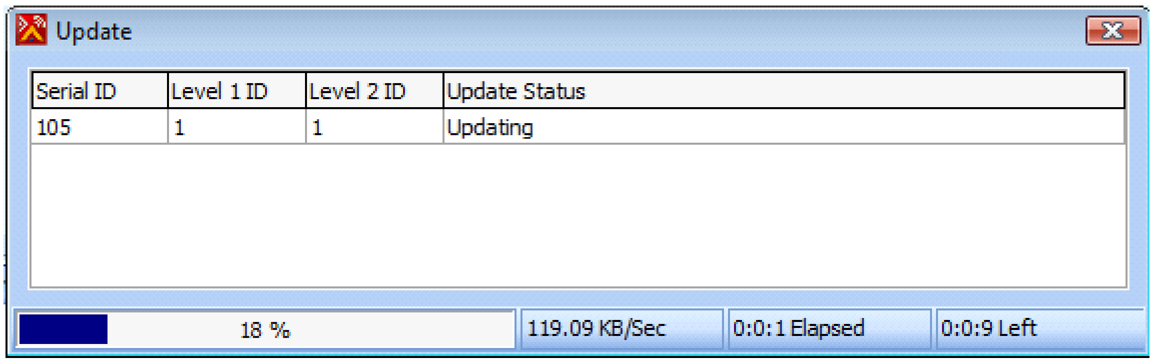


4. Select in the list your device, and then click on [OK].

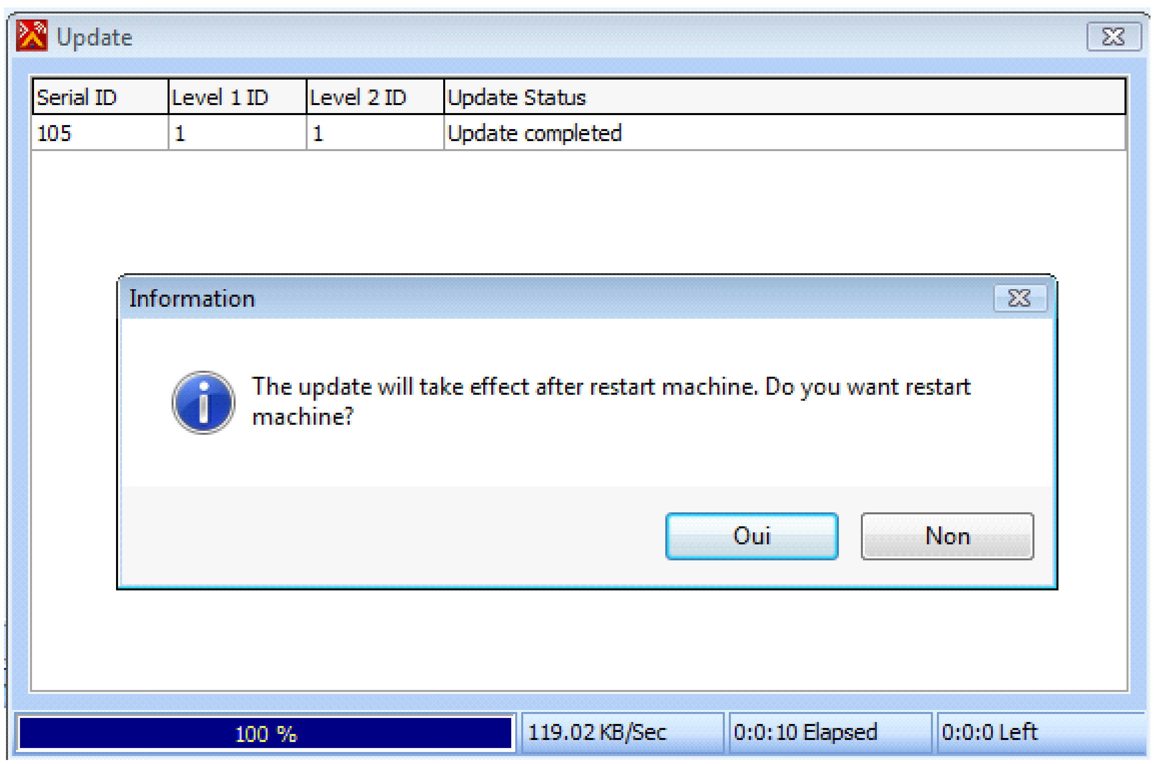
5. The browser open, search the ASU (Ateïs Studio Update) files on your PC. Then click [Open] to start the update process.



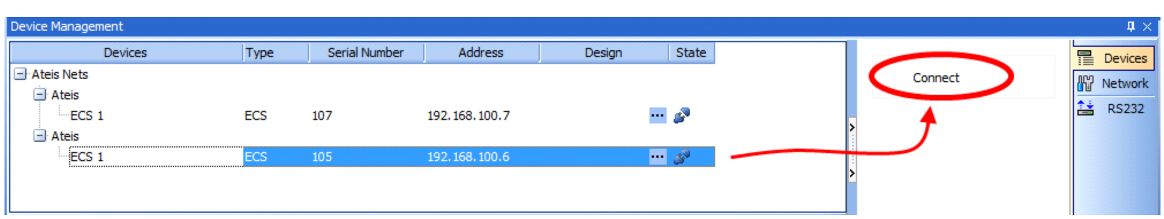
The Update window with its progress bar displays the state of the update:



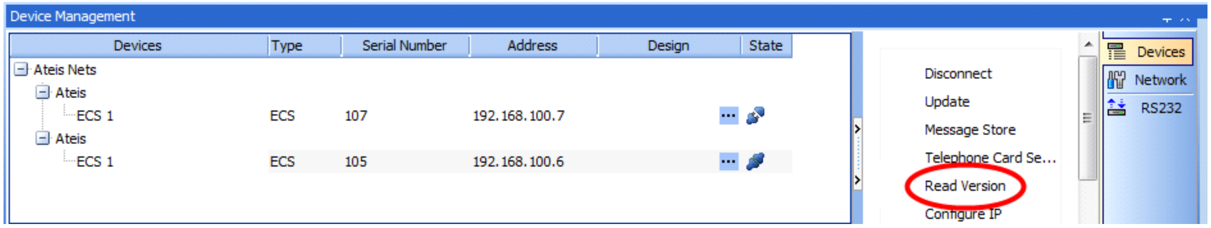
6. When the update is done, a pop-up window appears asking if you want the restart the device. This is necessary, so click on [Yes].

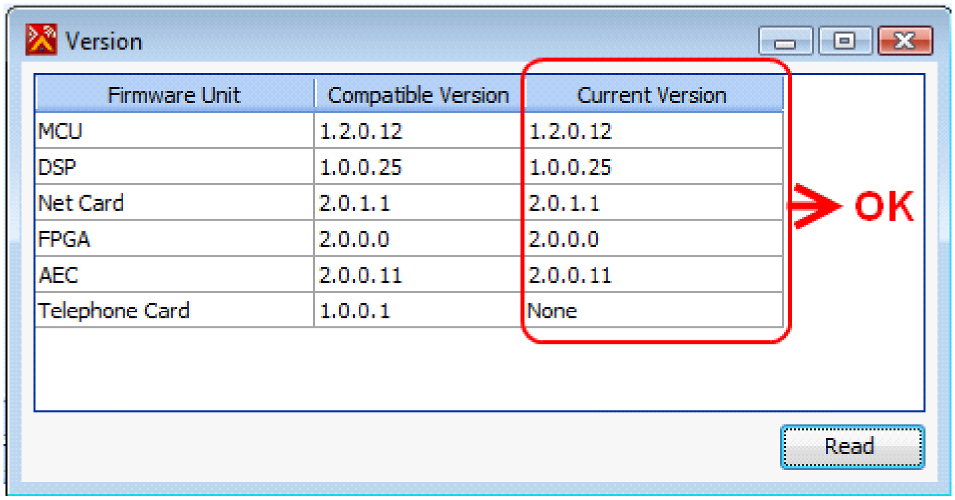


7. Now your device is updated. As the device is restarting, it is now unconnected. When the device is ready, then select it and connect it.



8. Our advice is to check the firmware version to be sure that they are now displayed in black, meaning that the device is ready for using.





5.4.4 Ateis Net Deploying

This function allow user to config networking state of devices on Ateis Net. When devices is first time setup, it need to assign an identification for each device, so you need to do a "config ip" for it. Follow the step listed below:

1. Open Device Management Window.

2. Search and Connect to the device.

3. Click [ATEIS NET Deploying] to execute the procedure.

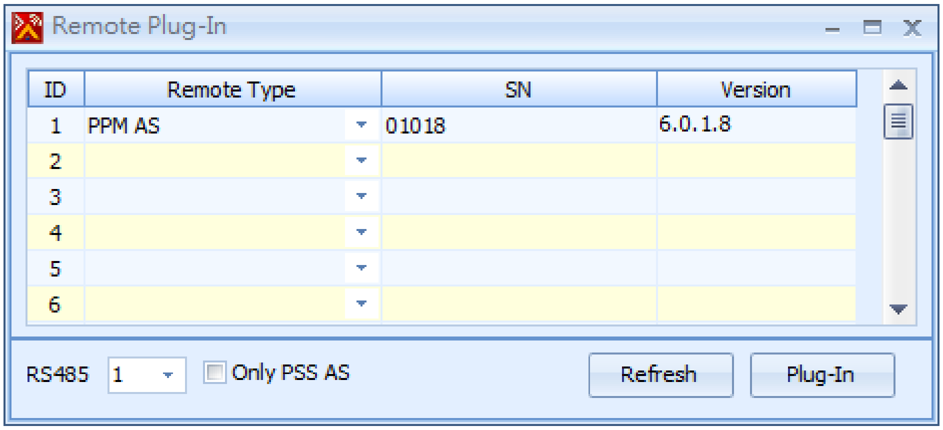
5.4.5 Remote Plugin

This function to register remotes to audio processor. When the remote is first time use, you need to do a "Remote Plug-in" to make audio processor know it. Follow the steps listed below:

1. Open Device Management Window.

2. Search and Connect to the device.

3. Click [Remote Plug-in] to open settings window.



- ID

  The internal identification for each remote devices.

- Remote Type

  To specify which type of remote device belong to the ID.

- SN

  Input serial number for the remote device.

- Version

  The version of remote device.

- RS485 Channel

  On left bottom a combo box is used to switch rs485 channel for settings.

- Option of Only PSS AS

  To specify the channel is connected with PSS AS. When you enable this option, it means no other remote devices are allowed connected to this channel.

- Refresh

  Read data from audio processor to refresh display information.

- Plug-In

  Set the editing data to audio processor.

4. Select Remote Type

   Select which type of remote device you'll using in the RS485 channel.

5. Input SN

   Type serial number of the remote device.

6. Press Plug-In Button

   Write settings to audio processor to register remote devices.

5.4.6 Reverse

This function allow user to reverse design from audio processors.

1. Open Device Management Window.

2. Search and Connect to the device.

3. Click [Reverse] to execute reverse procedure.

5.5 3rd Party Control

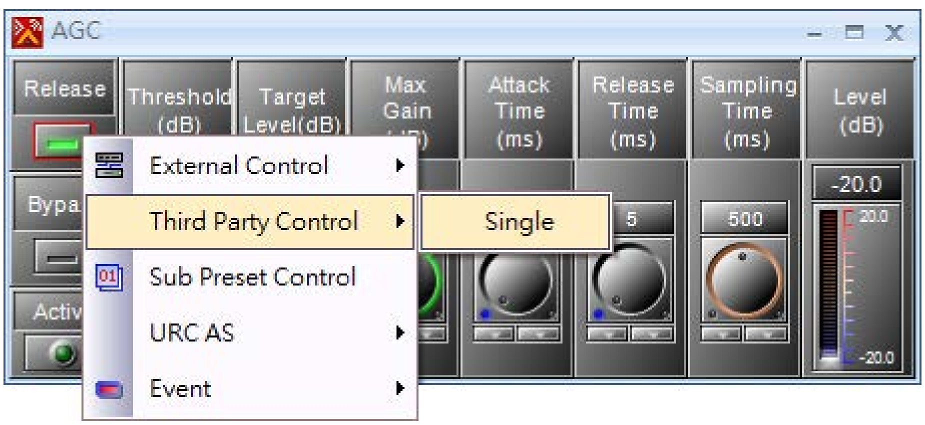
5.5.1 Ateis 3rd Party Control

5.5.1.1 Overview

Ateis device supplies a flexible way for controlling parameters inside it. Ateis 3rd party protocol allow external device to control Ateis device through Ethernet or RS232(depend on devices).

5.5.1.2 Assign Elements to 3rd Party Control

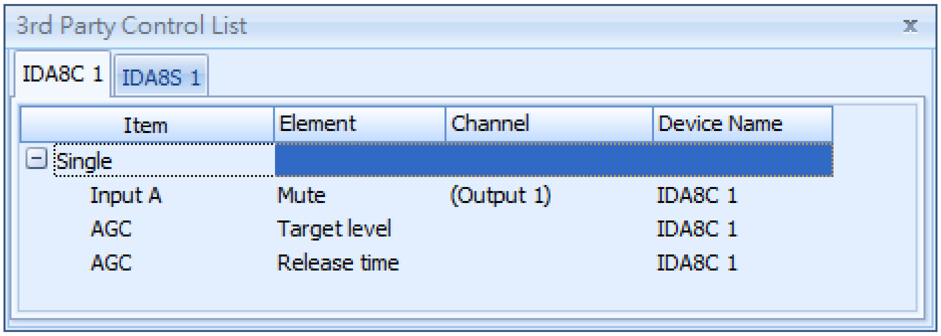
To make a element to be controlled by 3rd party protocol, you need to specify it is able to controlled by 3rd party devices, to do this setting, right click the element and click on menu [3rd Party Control > Single]



After the assignment is done, you can have a preview of the string to be used by 3rd party device for controlling Ateis devices. The 3rd Party Control Command is the preview window for the command.

To have a global view of all element controlled by 3rd party devices, you can use the 3rd Party Control List window. The window is described in later topic.

5.5.1.3 3rd Party Control List



This window list all elements available for 3rd party controlling. To open this window, click menu [View > 3rd Party Control List]

Followings are description for it:

- Tab for devices.

  There are separate tabs for each devices. In the tab, a grid list all elements controllable by 3rd party device.

- Item Field of Grid

  This show the elements set to 3rd party controlling by a tree structure. The parent item is single means the type is single. A single 3rd party control command controls only one element.

- Element Field of Grid

  The name of element listed.

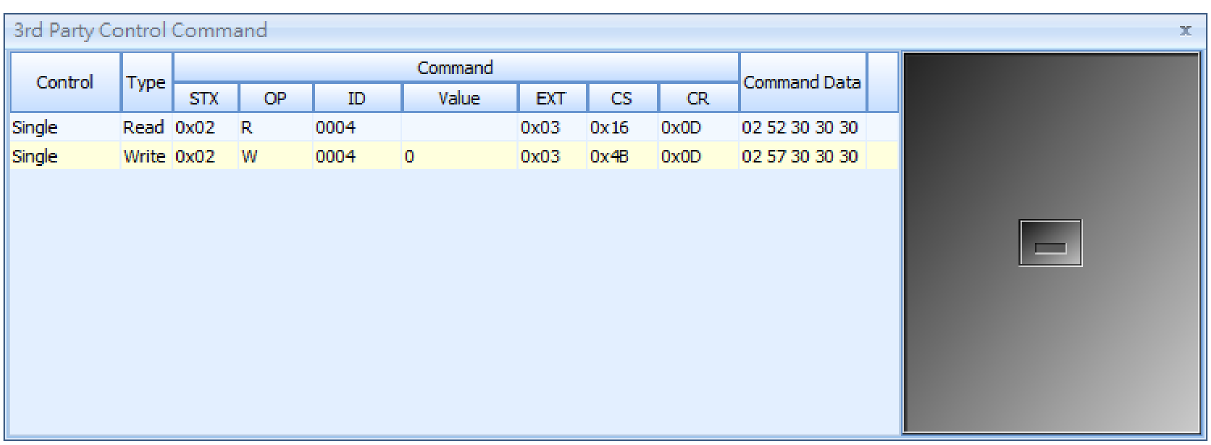
- Channel of Grid

  The channel in component element belonged.

- Device Name of Grid

  The component contains element listed.

5.5.1.4 3rd Party Control Command



This window can have a preview of the element that assigned for 3rd party control. In the left part, a grid

to provide information of command string for 3rd party device to control element. The right is a visual object allow user to change it's value to see the command string.

- Control

  Indicate the row of command string is for single control or multiple control.

- Type

  Show the command string is for reading or writing parameters of ateis devices.

- Command

  There are several sub-field in this field, STX, CP, ..., CR, each of them is part of a complete command string.

- Command Data

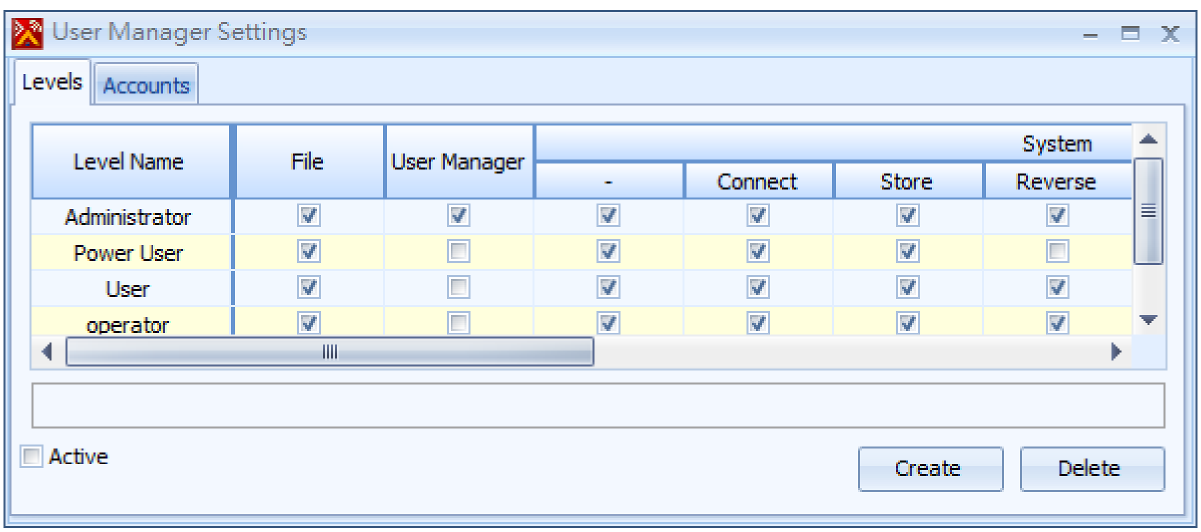
  The content of command string, represented by hexadecimal.

5.6 User Management

5.6.1 Overview

User management is a critical part of maintaining a secure system. Ineffective user and privilege management often lead many systems into being compromised. Therefore, it is important that you understand how you can protect your device through simple and effective user account management techniques. Ateis Studio provide a flexible user management mechanism to ensure the security for accessing system resources.

By clicking menu [View > User Management], you can open User Manager Settings window:



There are two pages in the window:

❖ User Levels

  An user level is a combination of user privileges, where a privilege is the right for executing some specific action. There are four default user levels to meet most applications with authority requirement.

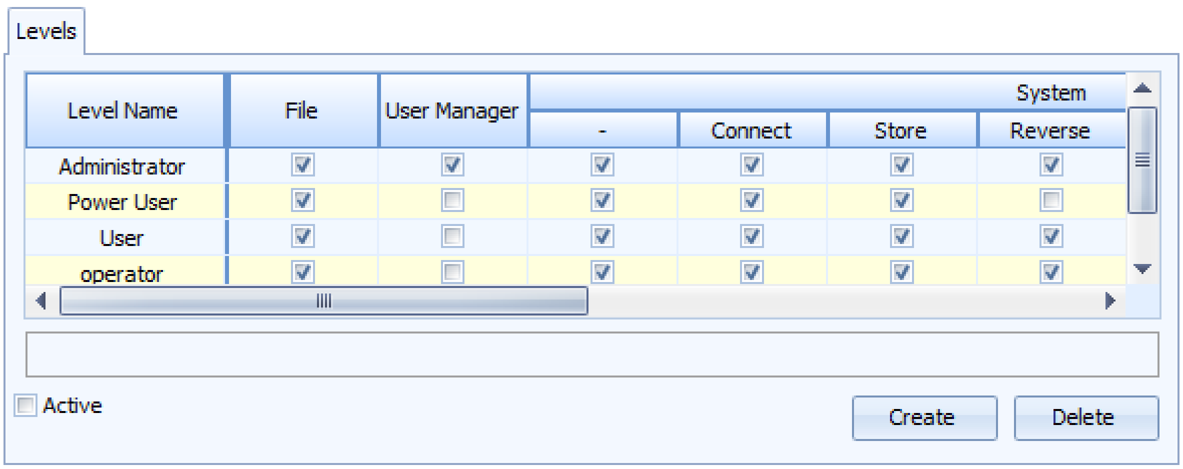
❖User Account

It allow you to create or remove users for controlling system. For each user, you can set it's user level by defaut or create a new level for customized privilege combination.

On left bottom, there is a check box [Active] to enable or disable user management for the configuration file.

5.6.2 User Levels

An user level is a combination of user privileges, where a privilege is the right for executing some specific action. There are four default user levels to meet most applications with authority requirement. Click page [Level] to edit user levels:



❖On top of window, there is a grid list all user levels including four default ones. Each column represent an action authority. Followings are description for each authority.

- File

  The right of opening configuration files.

- User Manager

  The right of edit user accounts and user levels.

- System(-)

  Select all authorities inside system.

- System(Connect)

  The right of connecting to Ateis devices.

- System(Store)

  The right of storing configuration to Ateis devices.

- System(Reverse)

  The right of reversing configuration to Ateis Studio from devices.

- System(Update)

The right of updating firmware of Ateis devices.

- System(Para. R/W)

  The right of reading or writing parameters in Ateis devices.

- System(Online)

  The right of online with Ateis devices.

- Master Preset(-)

  To select all authorities.in master preset.

- Master Preset(View)

  The right of viewing master preset configuration.

- Master Preset(Edit)

  The right of viewing editing settings belong to master presets.

- Master Preset(Control)

  The right of switching master presets.

❖ On the right bottom of window, there are two buttons:

- Create

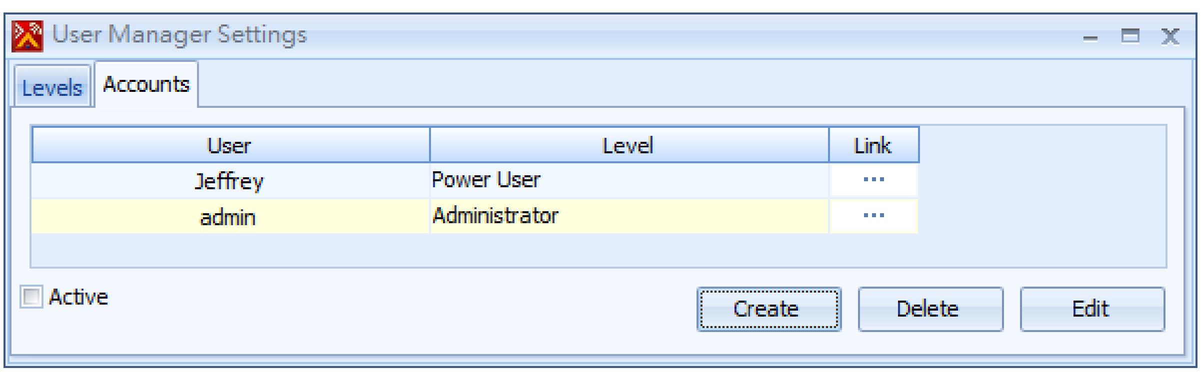
  To create a new user level, and enable desired authorities.

- Delete

  To remove select user level. Notes that Administrator, Power User and User is not allowed to remove.

5.6.3 User Accounts

Ateis device allow you to create multiple sers for controlling system. For each user you can specify different access right by giving an user level. By clicking page [Accounts], you can create or remove users.



❖ There is a grid to list all users, followings are description of each field:

- User

  The name of user.

- Level

  The level user belongs to.

- Link

  A ellipsis button to switch page to [Levels], and focus on the level that user belongs to.

❖ Three buttons on the right bottom of window:

- Create

  Create a new user.

- Delete

  Remove selected user.

- Edit

  Change password of selected user.

6 Components of Audio Processor

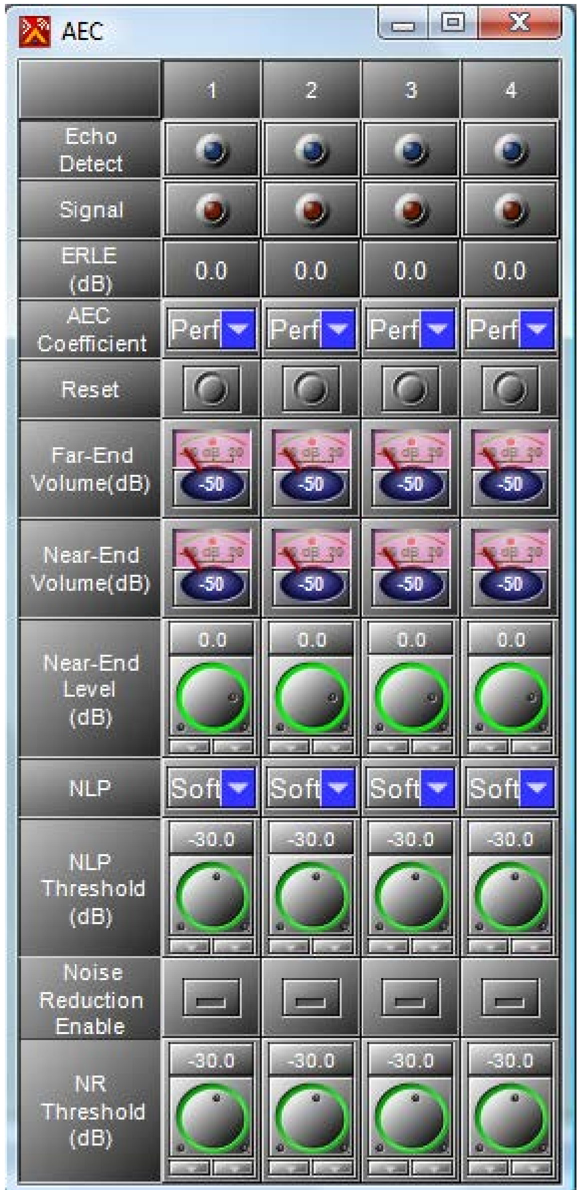
6.1 AEC

6.1.1 Parameters

This component is used to avoid reflected signals caused by acoustic echo between microphones and speakers, in remote conference applications.

The control window of the "AEC" module

is opened by a double click on the icon in the DESIGN AREA and appears like this:



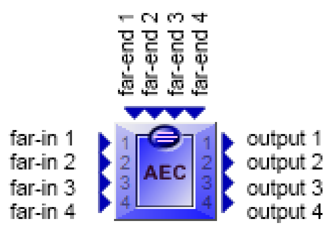
AEC Parameters

Parameter	Range	Comment
Echo Detect	led	This led lit when an echo is detected.
Signal	led	This led lit when an audio signal is detected.
ERLE (dB)		Echo Return Loss Enhancement. The decrement of echo, it indicates the quality of the AEC.
AEC Coefficient	Selector: Perform, Bypass, Hold	Function supplied for AEC controlled status. It offers 4 functions: • Perform: turn on the AEC function • Bypass: turn off the AEC function • Hold: hold the parameters of the echo coefficient
Reset	Button	Reset the echo cancellation on the channel

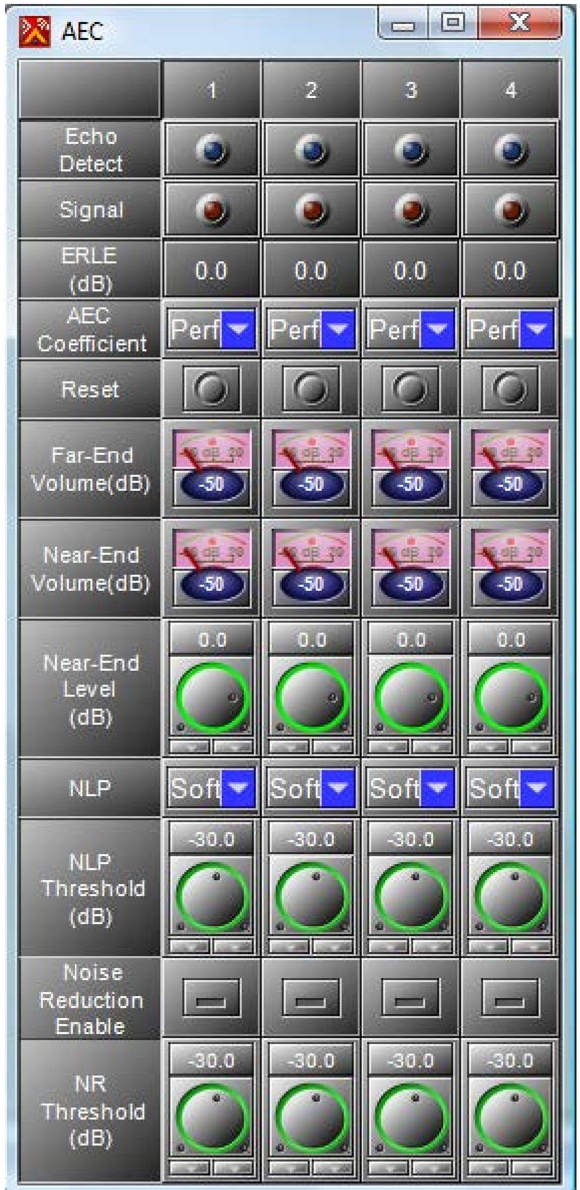
Parameter	Range	Comment
Far-End Volume	Vu-meter (dB)	Audio volume of the Far-end audio signal.
Near-End Volume (dB)	Vu-meter (dB)	Audio volume of the Near-end audio signal.
Near-End Level	-90 to 20 (dB)	volume adjustment of the near end audio signal
NLP	Selector: Soft, Medium, Aggressive, Off	NonLinear Process. The purpose is to suppress the remain echo in complex environment. It offers four functions: • off: NLP function off • soft: weak intensity of remain echo suppression. • medium: medium intensity of remain echo suppression. • aggressive: aggressive intensity of remain echo suppression.
NLP Threshold (dB)	-90 to 20 (dB)	The threshold value of the Nonlinear Process.
Noise Reduction Enable	Button	Enable the noise gate
NR Threshold	-90 to 20 (dB)	Threshold of the noise gate

6.1.2 Functions

The AEC component provides four fully independent channels.



On the component, each channel has one input called "far-in", one output, and one reference called "far-end". The far-end signal will be removed of the far-in signal, in order to send a proper signal to the other room.



❖AEC Coefficient / Reset

On each channel you can independently activate the echo cancellation process by setting the "AEC Coefficient" parameter on "Perform". You can also let it working without the process activates, by keeping the AEC coefficient by choosing "Hold". Of course, you can deactivate it on "Off".

The reset button is used to reset the AEC coefficient.

❖Volume display and control

The far-end volume and near-end volume vu-meters display respectively the far-end and near-end

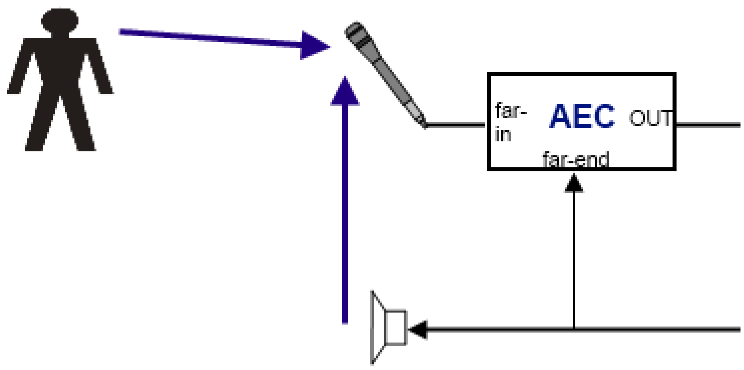
volume level.

If you want to modify the volume of a source (near-end inputs), then you have to use the "Near-End Level" knobs in the AEC component. When the system is properly adjusted, it is not advised to change the volume by using another parameter than this one.

❖ Non Linear Process (NLP)

The microphone captures the sound coming from the loudspeaker and also the speaker's voice.

If on the microphone capsule, the sound pressure level of the loudspeaker sound is near of the sound pressure level of the speaker's voice, then the AEC component won't be able to recognize them. In this case the AEC process is stopped and the non-linear process is activated.

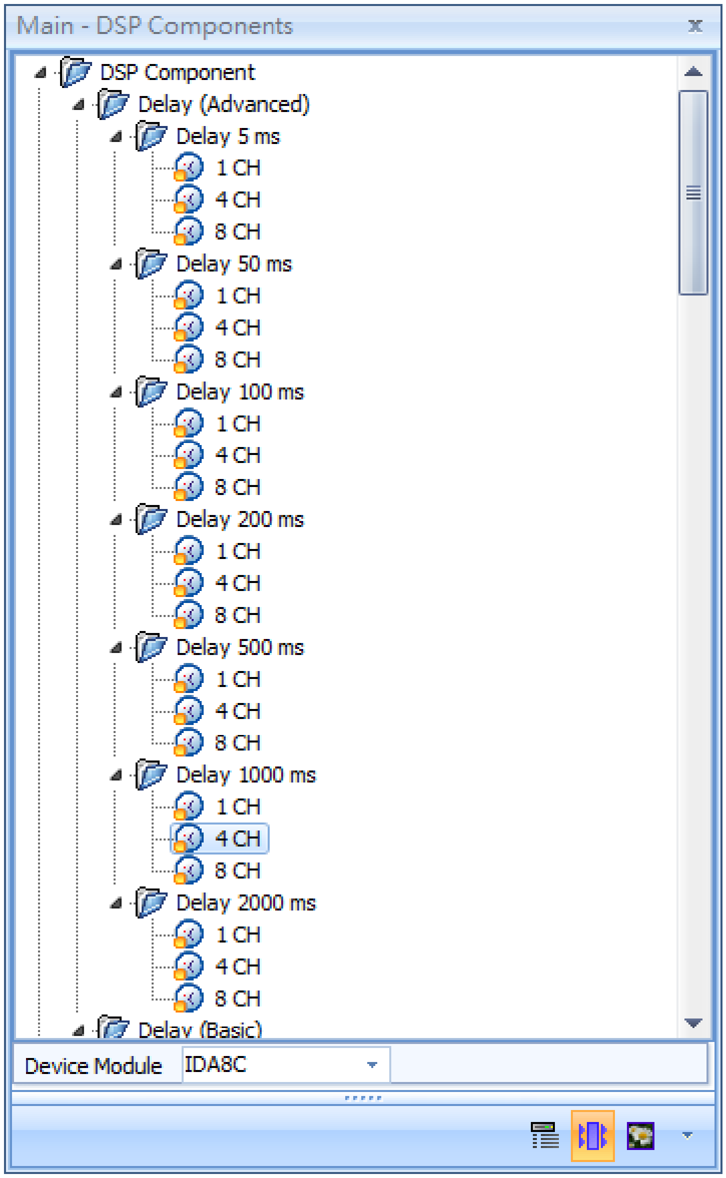


All you have to do is to choose the type of the non-linear process, soft, medium, aggressive or off.

The NLP threshold parameter is not available, the threshold is now automatically set.

6.2 Delay (Advanced)

❖Component Template

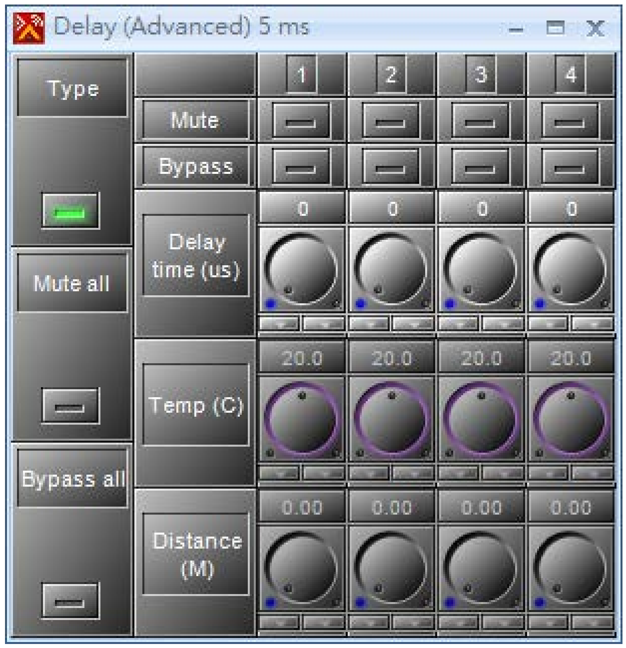


❖Component Appearance

❖Description

To delay the audio signal input to the component. Delay component record audio into a internal buffer, and then play back the stored audio based on parameters set by the user. There are two ways to adjust delay time for a channel. the first one is to adjust element Delay time, the other one is adjust Element Temp and Distance to compute delay time, Element Type is can switch between two modes.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Type	On	-	-	-	-
Mute all	Off	-	-	-	-
Bypass all	Off	-	-	-	-
Mute	Off	-	-	-	-
Bypass	Off	-	-	-	-
Delay time	0	0	*1	*2	μs
Temp	20	-10	50	0.1	℃
Distance	0	0	*3	0.01	m

*1: Maximum value of Delay time in various type of Delay (Advanced).

Type	Maximum
5ms	5000 μs
50ms	50000 μs
100ms	100000 μs
200ms	200000 μs
500ms	500000 μs
1000ms	1000000 μs
2000ms	2000000 μs

*2: Precision value of Delay time in various type of Delay (Advanced).

Type	Maximum
5ms	5 µs
50ms	20 µs
100ms	20 µs
200ms	20 µs
500ms	20 µs
1000ms	20 µs
2000ms	20 µs

*3: Maximum value of Distance is depend on the type of Delay (Advanced)

Type	Maximum
5ms	1.62 m
50ms	16.25 m
100ms	32.5 m
200ms	65 m
500ms	162.5 m
1000ms	325 m
2000ms	650 m

❖Element Description

- Type

  Choose between adjustments with Time or with Distance and temperature

- Mute all

  Mute all channel, the output of channel is mute if the value of this element or Mute is on.

- Bypass all

  Bypass audio signal of all channel, i.e. disable the delay function.

- Mute

  Mute the channel. the output of channel is mute if the value of this element or Mute all is on.

- Bypass

  Bypass audio signal of the channel.

- Delay time

  Specify how much time will be delayed for the audio signal of channel.

- Temp

  Specify the temperate to compute the delay time for channel.

- Distance

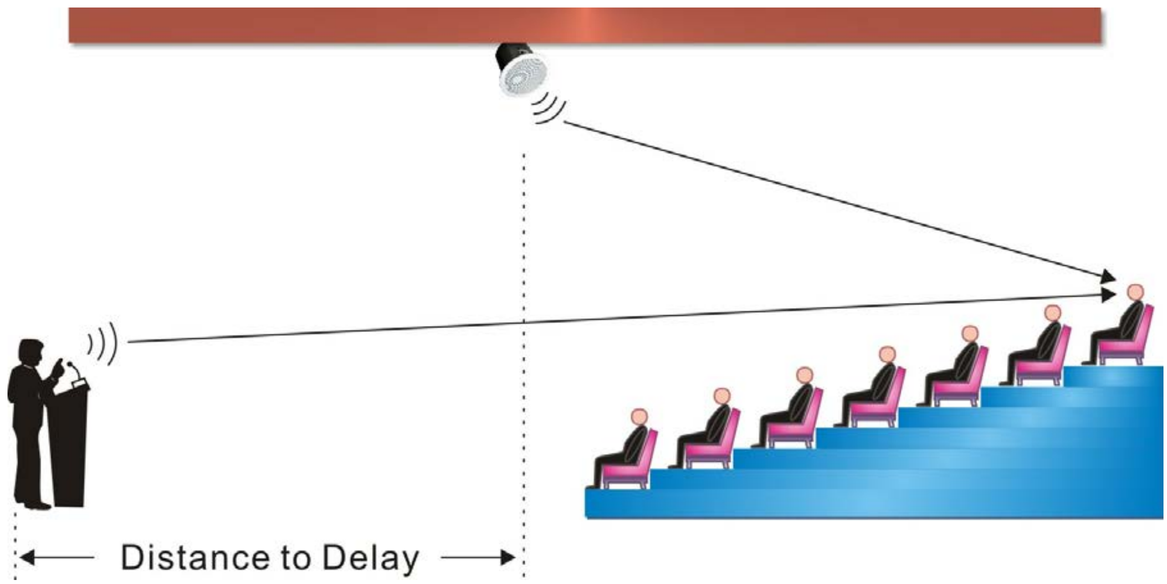
  Specify the Distance to compute the delay time for channel.

Note: If a channel is mute and bypass at the same time, the audio signal of the channel will mute.

❖Application

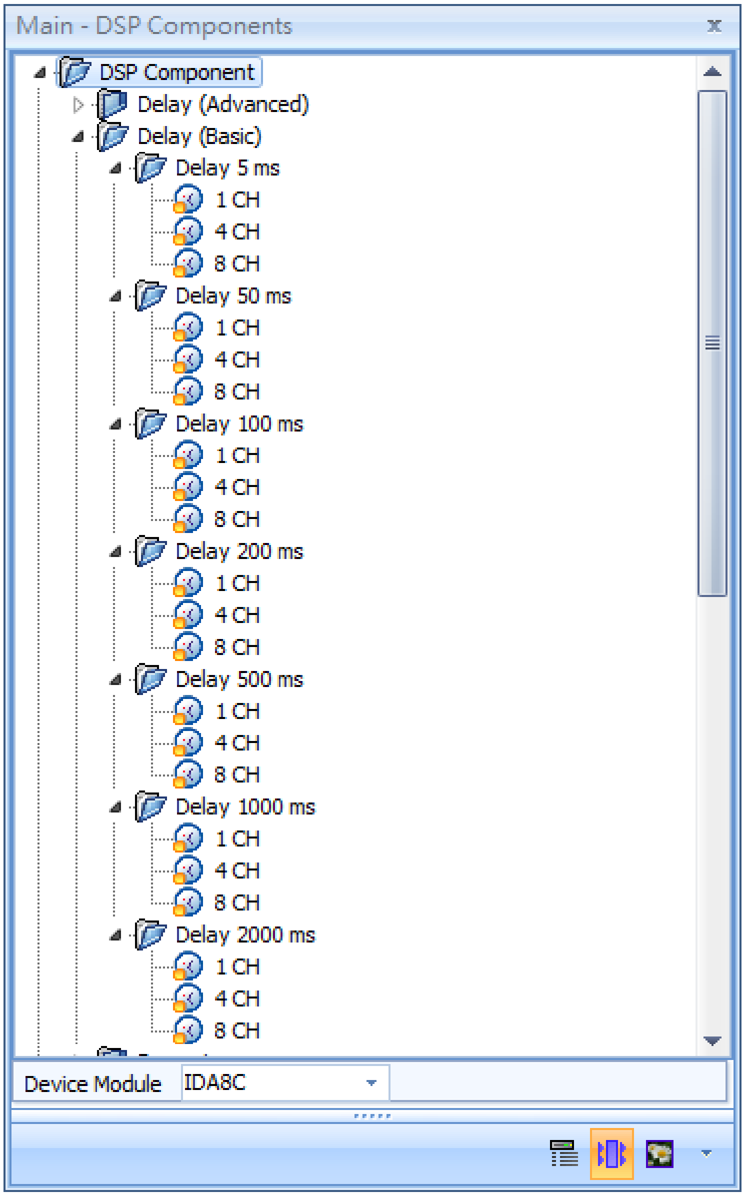
Imagine a conference in a large room. Normally speakers are beside the people speaking, so listeners at the end of the room receive too weak sound. Solution is to put also speakers in the

middle of the room. but in this case listeners will receive sound from speakers with time difference due to to distance between speakers.Solution is to add delay to speakers placed in the middle of the room. delay value compensates the distance.

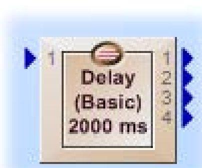


6.3 Delay (Basic)

❖ Component Template



❖ Component Appearance

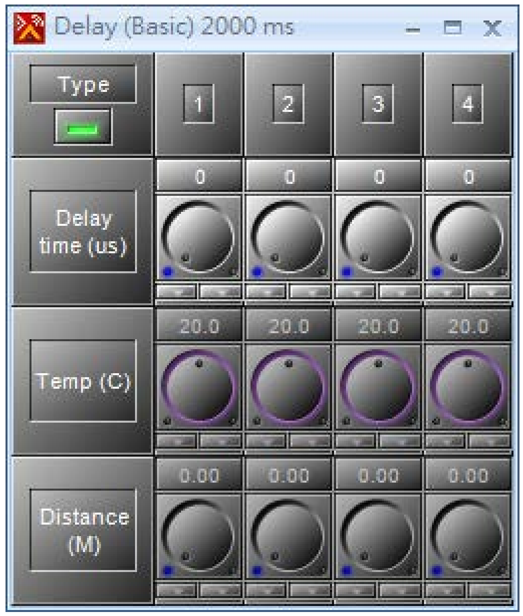


❖ Description

To delay the audio signal input to the component. This component is an compact version of Delay

(Advanced) that is not allowed to mute or bypass the signal of input. There are two ways to adjust delay time for a channel. the first one is to adjust element Delay time, the other one is adjust Element Temp and Distance to compute delay time, Element Type is can switch between two modes.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Type	On	-	-	-	-
Delay time	0	0	*1	*2	µs
Temp	20	-10	50	0.1	℃
Distance	0	0	*3	0.01	m

*1: Maximum value of Delay time in various type of Delay (Basic).

Type	Maximum
5ms	5000 µs
50ms	50000 µs
100ms	100000 µs
200ms	200000 µs
500ms	500000 µs
1000ms	1000000 µs
2000ms	2000000 µs

*2: Precision value of Delay time in various type of Delay (Basic).

Type	Maximum
5ms	5 µs
50ms	20 µs
100ms	20 µs
200ms	20 µs
500ms	20 µs

Type	Maximum
1000ms	20 µs
2000ms	20 µs

*3: Maximum value of Distance is depend on the type of Delay (Basic)

Type	Maximum
5ms	1.62 m
50ms	16.25 m
100ms	32.5 m
200ms	65 m
500ms	162.5 m
1000ms	325 m
2000ms	650 m

❖ Element Description

- Type

  Choose between adjustments with Time or with Distance and temperature

- Delay time

  Specify how much time will be delayed for the audio signal of channel.

- Temp

  Specify the temperate to compute the delay time for channel.

- Distance

  Specify the Distance to compute the delay time for channel.

Note: If a channel is mute and bypass at the same time, the audio signal of the channel will mute.

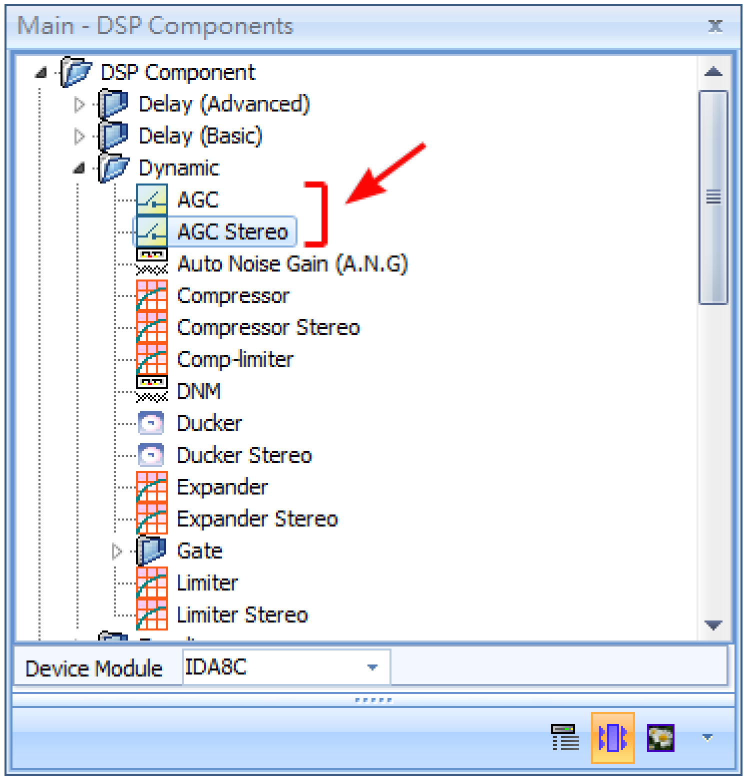
❖ Application

Refer to Delay (Advanced).

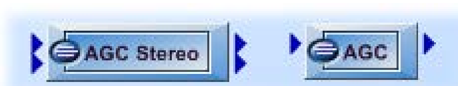
6.4 Dynamic

6.4.1 AGC\AGC Stereo

❖ Component Template



❖ Component Appearance

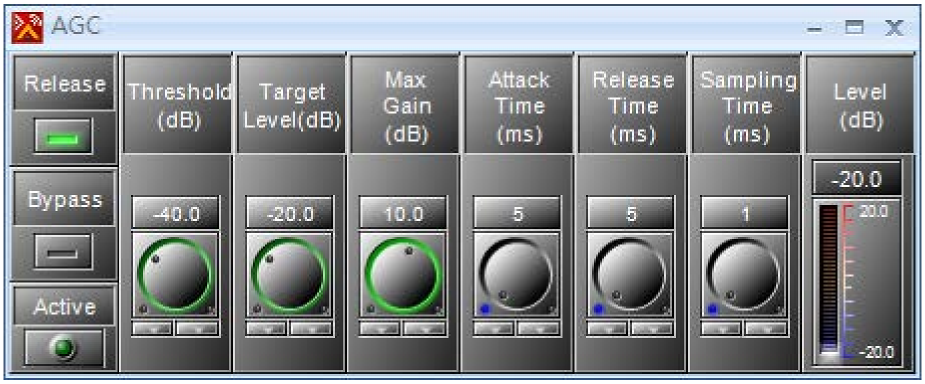


❖ Description

With this Automatic Gain Control, the input signal can be increase or decrease to a target level. For example, without AGC the sound emitted from an AM radio receiver would vary to an extreme extent from a weak to a strong signal; The AGC effectively reduces the volume if the signal is strong and raises it when it is weaker.

you can adjust automatically the gain of the source by setting a target level. There two type of AGC, one is signal channel, the other one is stereo.

❖ Control Window



The control window of AGC Stereo the same as AGC's.

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Release	On	-	-	-	-
Bypass	Off	-	-	-	-
Active	Off	-	-	-	-
Threshold	-40.0	-60.0	0	0.1	dB
Target Level	-20.0	-35.0	10	0.1	dB
Max Gain	10.0	0.0	18	0.1	dB
Attack Time	5	1	1K	1	ms
Release Time	5	1	1K	1	ms
Sampling Time	500	1	1K	1	ms
Level	-20.0	-20.0	20	0.1	dB

The element properties of AGC Stereo is the same as AGC's.

❖Element Description

- Release

  If the release button is on, the release time will be the element Release Time. Otherwise, the release time will be 10 ms.

- Bypass

  Disable AGC Function, i.e. bypass audio signal from input to output.

- Active

  Flash when the gain is controlled (AGC is active).

- Threshold

  Sets the threshold level above which the gain will be controlled.

  Don't set a too much low level, otherwise it will hear some unexpected little sound (breath, ambient noise…).

- Target Level

  Determine which relative volume you want to target.

- Max Gain

Determine the maximum of automatic gain.

⚠ Limit the max gain to keep a natural audio dynamic.

- Attack Time

  Adjust the fade in time needed to reach the target level.

  ⚠ A short time will give an uncomfortable hearing, with audio peak when the speaker begins to talk.

- Release Time

  Adjust the fade out time of the signal when AGC is no longer working. This value is ignored if the "Release button" is not pushed.

  ⚠ A short time will give an uncomfortable hearing, with audio slack between the speaker's words.

- Sampling Time

  Average time needed by measurement input level.

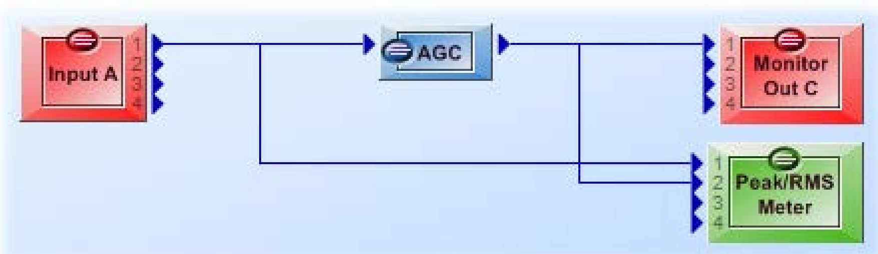
  ⚠ A too long time will cause the AGC become insensitive on short peaks.

- Level

  Level of the gain that AGC increase or decrease the input signal.

❖ Example

As example, we want to automatically adjust the gain of a kind of vocal microphone.

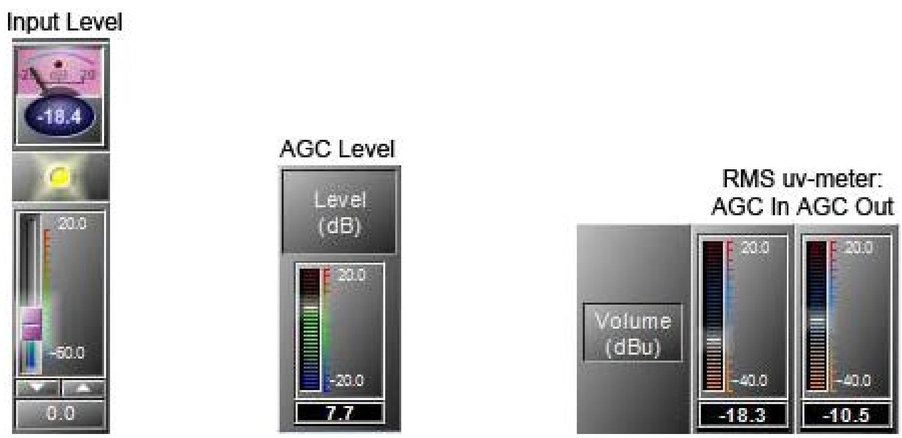


On the "Input A" component, the input sensitivity is set to -40dB for standard dynamic microphone.

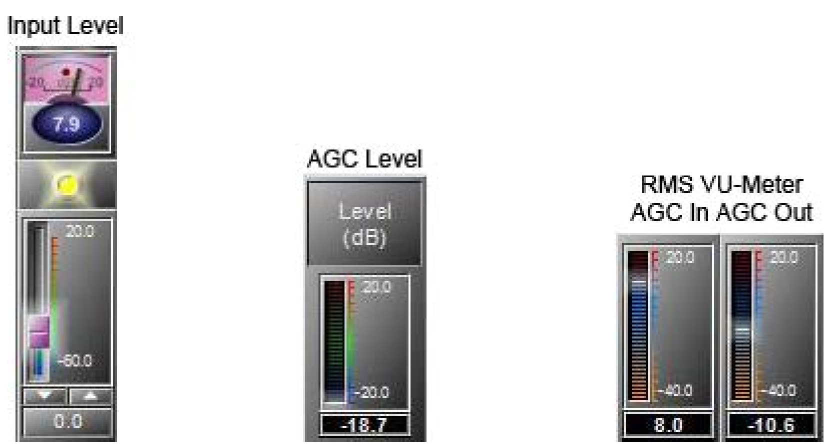
Threshold: Set at -25dB. The level is adjusted only when the input signal is above -25dB. This is to keep a natural dynamic.

- Target level: We want a final target level at -10dB.
- Maximum gain: Set at 10dB, because giving more gain will generate some uncomfortable level differences.
- Release time: Set at 800(ms), enough time to avoid hard transition when the speaker talks again, after a little break.
- Attack time: Set at 50(ms), enough time to avoid hard transition when the speaker begin to talks, and not too long because we want a stable level quickly (not since the middle of the sentence).
- AVG time: Set at 200(ms), not too long, to detect short peaks in the speech and to be responsive. Not too short to avoid level change too often.

You can see on the vu-meter that when the speaker talks at -18.3 dBu, the final level is around -10.5 dBu:

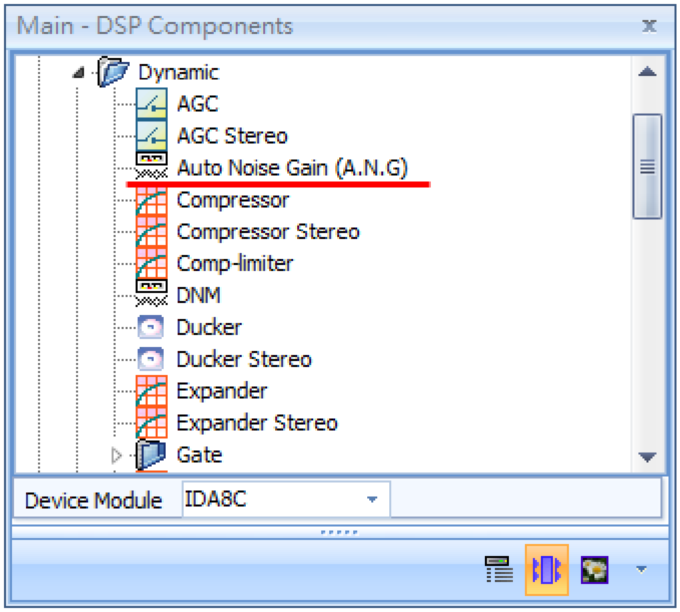


You can see on the vu-meter that when the speaker talks at 8 dBu, the final level is around -10.6 dBu:



6.4.2 Auto Noise Gain (A.N.G)

❖Component Template



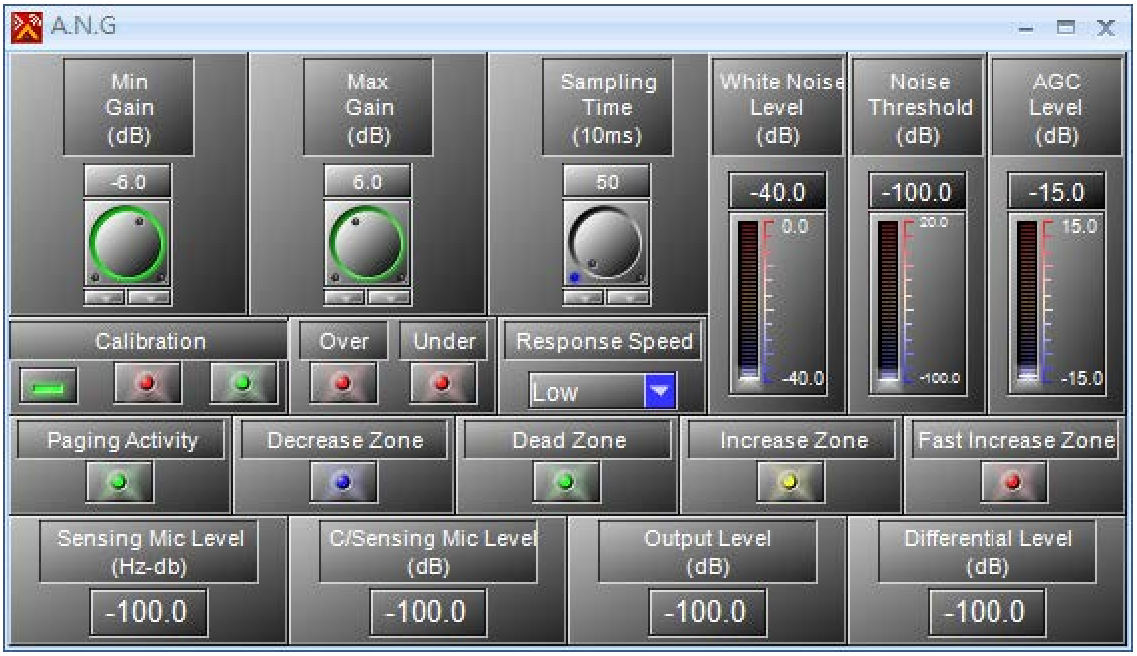
❖Component Appearance

- S.M: Sensing microphone input
- P.M: Paging input

❖Description

Adjust the audio level of the paging input (P.M.) depending of the ambient noise. The ambient noise is measured with a sensing microphone connected to the sensing microphone input (S.M.). Our advice is to use the noise sensing microphone NSM (Ateïs), but you can also use another one.

❖Control Window



❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Min Gain	-6.0	-15.0	0	0.1	dB
Max Gain	6.0	0	15.0	0.1	dB
Sampling Time	50	50	60000	50	10ms
White Noise Level	-40.0	-40.0	0	0.1	dB
Noise Threshold	-100.0	-100.0	20.0	0.1	dB
AGC Level	-15.0	-15.0	15.0	0.1	dB
Calibration	Off	-	-	-	-
Calibration Proceeding	Off	-	-	-	-
Calibration Finish	Off	-	-	-	-
Over	Off	-	-	-	-
Under	Off	-	-	-	-
Response Speed	Low	-	-	-	-
Paging Activity	Off	-	-	-	-
Decrease Zone	Off	-	-	-	-
Dead Zone	Off	-	-	-	-
Increase Zone	Off	-	-	-	-
Fast Increase Zone	Off	-	-	-	-
Sensing Mic Level	-100.0	-100	200	0.1	Hz-dB
C/Sensing Mic Level	-100.0	-100	200	0.1	dB
Output Level	-100.0	-100	200	0.1	dB
Differential Level	-100.0	-100	200	0.1	dB

Item list of Response Speed:

No.	Name
1	Low
2	Medium
3	High

❖Element Description

- Min Gain

  The minimum level compensation.

- Max Gain

  The maximum level compensation.

- Sampling Time

  Determine the period time between two ambient noise measures. The less value is more sensitivity, it means the compensation gain of paging input is response more quickly of environment noise.

- White Noise Level

  White noise is used for calibration, the level is automatically adjusted in calibration process.

- Noise Threshold

  The level of background noise measured when calibration button pressed.

- AGC Level

  Display the current compensation gain.

- Calibration

  Click on to start the automatic calibration procedure.

- Calibration Proceeding

  Red led, light on if the calibration procedure is proceeding.

- Calibration Finish

  Green led, light on if the calibration procedure is success.

- Over

  Indicates the calibration result:

  The white noise level needed is over the range. A pop-up window will shows you what to do.

- Under

  Indicates the calibration result:

  The white noise level needed is under the range.A pop-up window will show you what to do.

- Response Speed

  Select here the reaction speed of the A.N.G.

That's means the speed to adapt the P.M. audio level depending of the ambient noise.

- Paging Activity

  Light on when the paging is active.

- Decrease Zone

  Indicates that the AGC gain is decreasing.

- Dead Zone

  Indicates that the AGC is not changing.

- Increase Zone

  Indicates that the AGC gain is increasing.

- Fast Increase Zone

  Indicates that the AGC gain is increasing fast.

- Sensing Mic Level

  Show the input level of the sensing microphone.

- C/Sensing Mic Level

  Show the input level of the sensing microphone through the internal calibration transfer function.

- Output Level

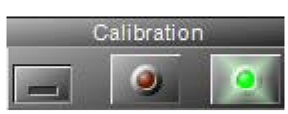
  Show the output level.

- Differential Level

  Difference between the Output level and C/Sensing Mic level.

❖ Procedure to set the A.N.G

1. Connect the sensing microphone to the S.M. input, and the paging audio source to the P.M. input. The NSM sensing microphone needs an input sensitivity of 0dB (software). The max cable length between the NSM and the audio processor is 500 meter with CAT-5 cable.
2. Link the output of the ANG component to an Output component where is connected a loudspeaker.
3. Click on the Calibrate button, to start the calibration procedure. The system will start to estimate the acoustic transfer function between loudspeakers and microphone when user clicks the Calibration button, and it will automatically stop calibrating until the system find a better transfer function.
4. The calibration red led light up, means the calibration procedure is running. During the procedure, don't move the physical installation, and don't make punctual noise. The calibration must be done in normal ambient conditions.
5. Wait until the end of the calibration. The red calibration led light off, and the green calibration led light up.

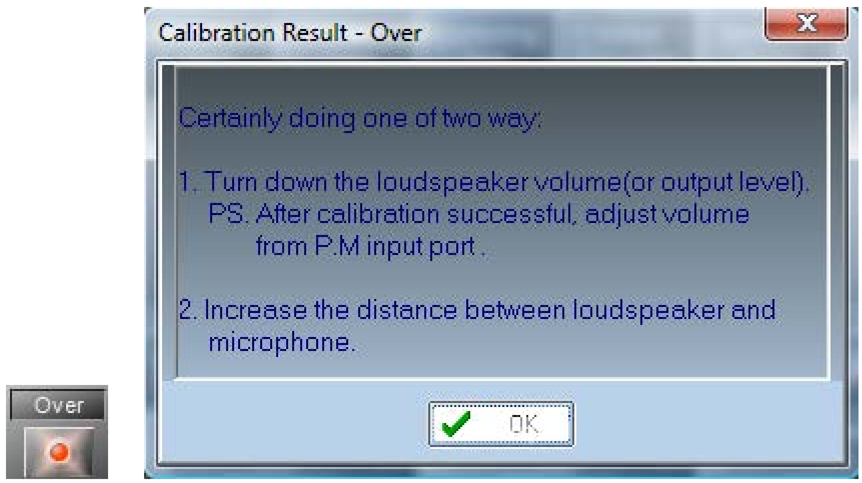


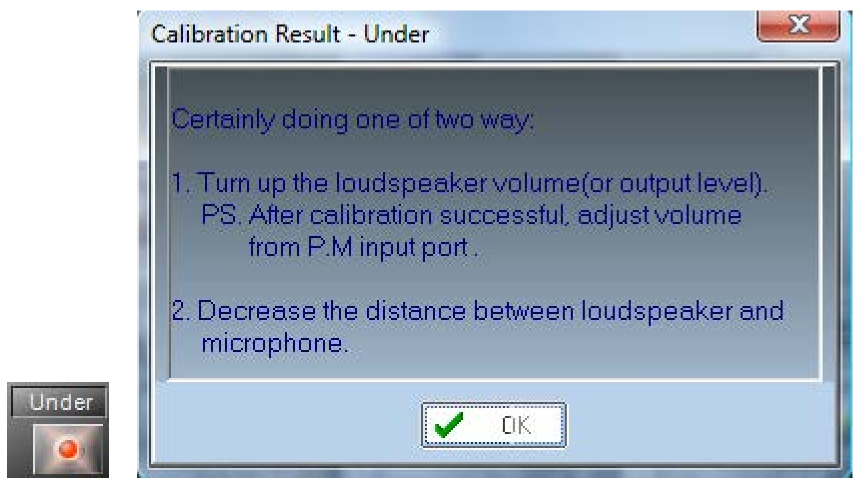
6. Now look the Over/Under red led:

   -If both Over/Under led are OFF, it means the calibration has properly finished and the ANG is ready.

   -If one of the Over/Under led light ON, it means the white noise level needed is out of range.

7. Depending of the issue, a pop-up window will show you what to do. Follow the most usefully advice and restart the calibration procedure until the both Over/Under led are OFF after the calibration.





8. The system is properly set.

Even when the paging is not active, the ANG is still adjusting the AGC level with the ambient noise to be ready when the paging will begin.

Notice:

- The calibration should be done in normal ambient noise (shouldn't have suddenly loud noise during calibration).
- Do not adjust the gain of output, amplifier or sensing microphone during or after calibration. (You can adjust volume of the audio source connected to P.M input.
- Theoretically, when the calibration has finished, the AGC level will be stable around 0 dB in the environment of no increasing noise. But every calibration might exist deviation, if the AGC

level is not stable around0 dB, user could adjust the gain of output function, amplifier or sensing microphone to let AGC level stable around 0 dB.

- When the system is calibrating, clicking on the calibration button will cancel the calibration at that moment. (The system will use the calibration result which was successfully calibrated at last time.)
- After calibration successful, the paging microphone shouldn't be placed nearly to sensing microphone.(because the user's voice will be also received into sensing microphone.)
- AGC level is still calculating even in inactivate mode.The level is depend on background noise. It will support a good initial gain for paging level.

❖ Use the A.N.G

When the calibration green led lit, the system is ready. In our example, the A.N.G has detected an ambient noise average of -12dB. We choose a medium "response speed" and a short "sampling time" to be enough reactive quickly.

Now the A.N.G will automatically adjust the level on the (P.M) signal depending on the ambient noise detected on the sensing microphone. The A.N.G will working when the paging activity led will light, meaning that an audio signal is detected on the (P.M) input.

Dead Zone:

If the sensing microphone doesn't detect more noise than during the calibration, the A.N.G works in the "Dead Zone" which means that there is no need to adjust the gain.

As the noise level is near than during the calibration, the differential level is almost null. You can see the AGC level near from 0 dB, meaning no compensation.

Increase Zone:

If the sensing microphone detects more noise than during the calibration, the A.N.G works in the "Increase Zone" which means that the gain must be increased to compensate the ambient noise.

In our example, the noise is now louder than during the calibration. The differential level is around 4.4dB. You can see the AGC level at 3,5dB, meaning the (P.M) signal is increased of 3,5dB. As soon as the AGC level reach the level needed to compensate the ambient noise (or the Max Gain), the A.N.G will goes back in the "Dead Zone".

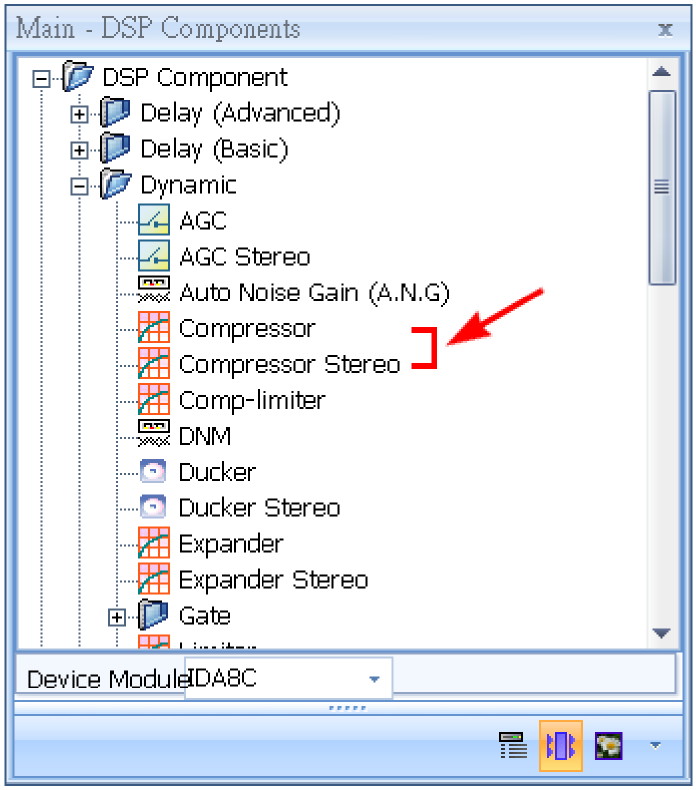
Fast Increase Zone:

If the sensing microphone detects more noise than during the calibration, the A.N.G works in the

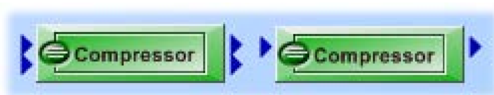
"Fast Increase Zone" which means that the gain must be quickly increased to compensate the loud ambient noise.

6.4.3 Compressor\Compressor Stereo

❖ Component Template



❖ Component Appearance



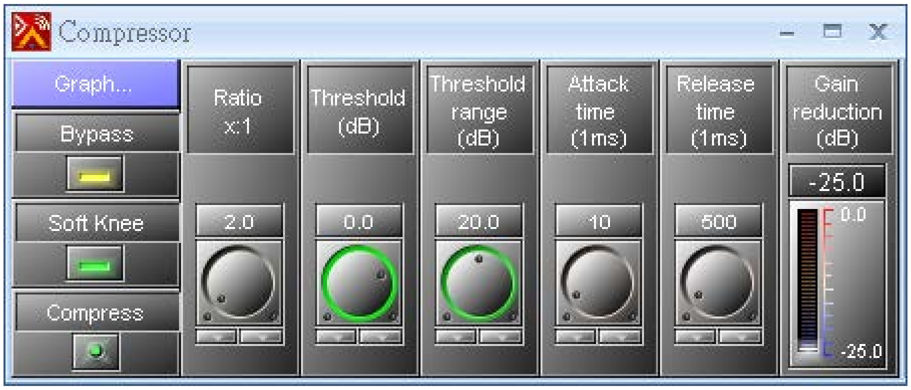
❖ Description

A Compressor is an automatic volume control. Loud sounds over a certain threshold are reduced in level while quiet sounds remain untreated. In this way it reduces the dynamic range of an audio signal. This may be done for aesthetic reasons, to deal with technical limitations of audio equipment, or to improve audibility of audio in noisy environments.

Compression reduces the level of the loud sounds, but not the quiet sounds, thus, the level can be raised to a point where the quiet sounds are more audible without the loud sounds being too loud.

Compressor Stereo is the same that the Compressor component, but this works on the both stereo channels. The detection is done on the both input signals. The same compression is applied on the both output signals.

❖ Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Soft Knee	On	-	-	-	-
Compress	Off	-	-	-	-
Ratio	2.0	1.1	20.0	0.1	-
Threshold	0.0	-60.0	20.0	0.1	dB
Threshold Range	10.3	0	40.0	0.1	dB
Attack Time	10	1	100	1	ms
Release Time	500	100	5000	1	ms
Gain Reduction	-25.0	-25.0	0	0.1	dB

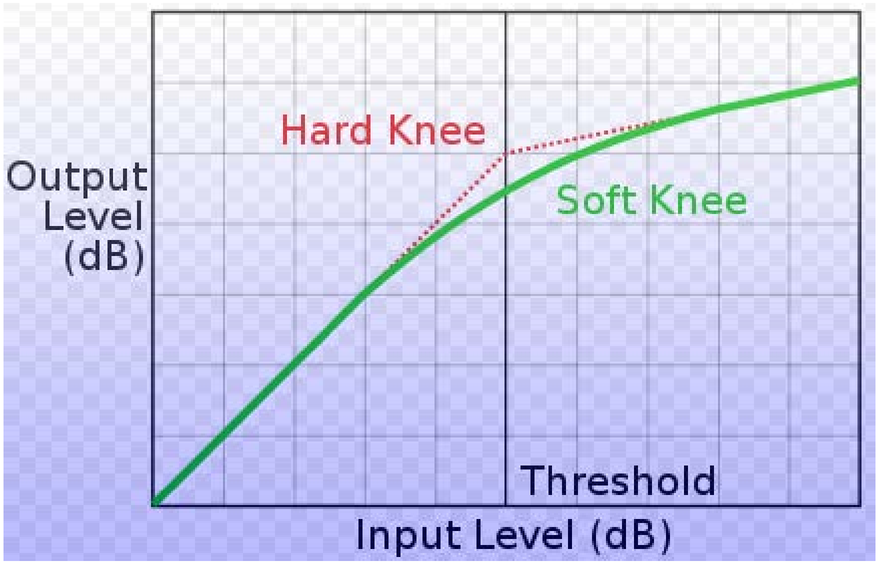
❖Element Description

- Bypass

  Bypasses the compressor's function.

- Soft Knee

  Enables/disables the “Soft Knee” mode. This element controls whether the bend in the response curve is a sharp angle or has a rounded edge. A soft knee slowly increases the compression ratio as the level increases and eventually reaches the compression ratio set by the user. A soft knee reduces the audible change from uncompressed to compressed, especially for higher ratios where the changeover is more noticeable.



- Compress

  Indicates compression.

- Ratio

  Sets compression ratio.

- Threshold

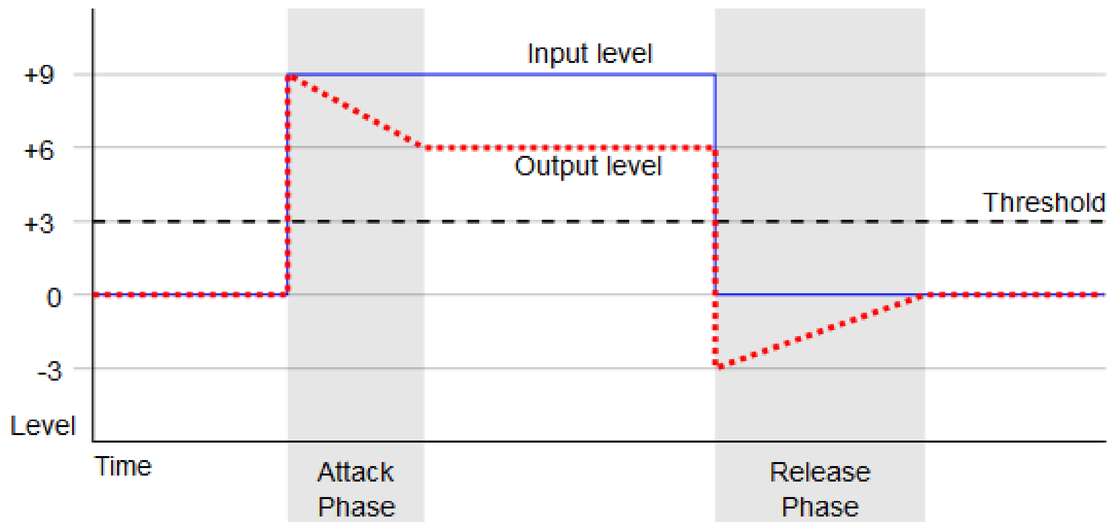
  Sets the threshold level above which compression begins.

- Threshold Range

  Sets the range of level, in which the ratio gradually alters from 1:1 to the value set by the ratio parameter (soft knee function).

- Attack Time

  Sets the time it takes to respond to the input signal. A compressor might provide a degree of control over how quickly it acts. The 'attack phase' is the period when the compressor is decreasing gain to reach the level that is determined by the ratio.



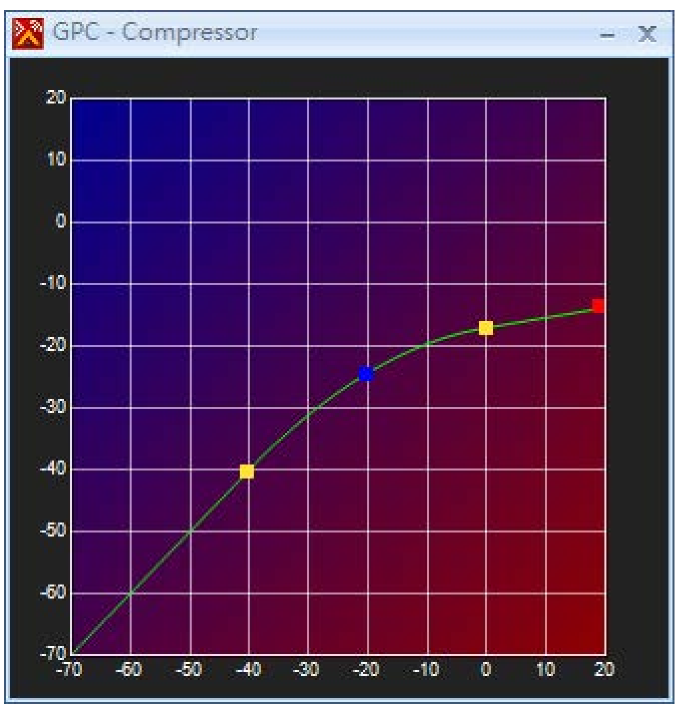
- Release Time

  Sets the time it takes to release gain reduction. The 'release phase' is the period when the compressor is increasing gain to the level determined by the ratio, or, to zero dB, once the level has fallen below the threshold.

- Gain Reduction

  Reflects the current amount of gain reduction.

❖ Graphical Control Window



- Axis
  - X: Input signal level(dB)

  - Y: Output signal level(dB)

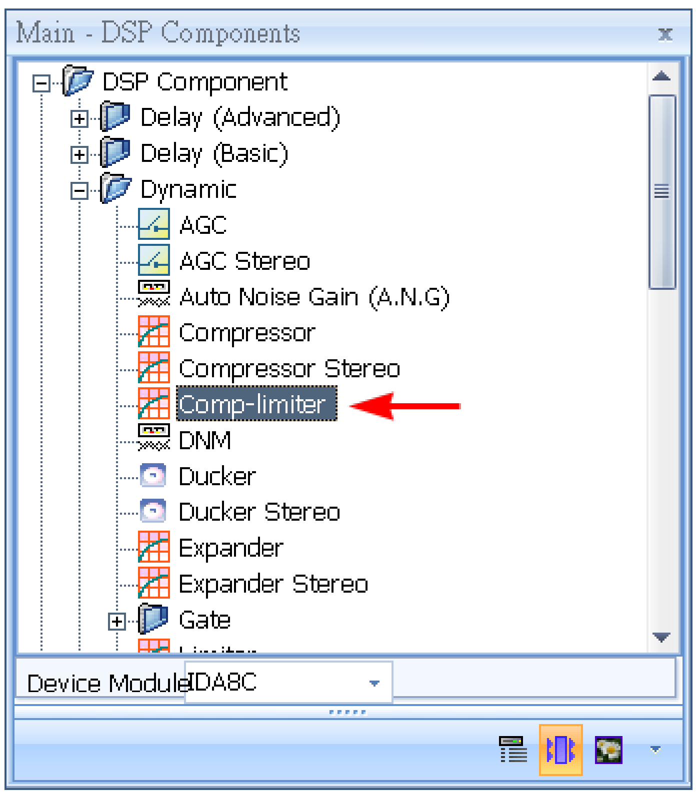
- Control Points
  - Blue: Adjust Threshold.
  - Red: Adjust Threshold Range.
  - Yellow: Adjust Ratio

❖Application

Engineers wishing to achieve dynamic range reduction with few obvious effects might choose a relatively low threshold and low compression ratio so that the source material is being compressed very slightly most of the time. To deliberately soften the attack of a snare drum, they might choose a fast attack time and a moderately fast release time combined with a higher threshold. To accentuate the attack of the snare, they might choose a slower attack time to avoid affecting the initial transient.

6.4.4 Comp-limiter

❖Component Template



❖Component Appearance

❖Description

Comp-limiter is a combination of the Compressor and the Limiter Components.

Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Soft Knee	On	-	-	-	-
Limiting (Limiter)	Off	-	-	-	-
Hard Clipping (Limiter)	Off	-	-	-	-
Threshold (Limiter)	0	0	20.0	0.1	dB
Release Time (Limiter)	500	100	5000	1	ms
Compress (Compressor)	Off	-	-	-	-
Ratio (Compressor)	2.0	1.0	20.0	0.1	-
Threshold (Compressor)	-10.0	-60.0	0	0.1	dB
Release Time (Compressor)	500	100	5000	1	ms
Attack Time (Compressor)	10	1	100	1	ms
Gain Reduction	0	-25.0	0	0.1	ms

❖Element Description

- Bypass

  Bypasses the Comp-limiter's function.

- Soft Knee

  Enables/disables the “Soft Knee” mode.

- Limiting(Limiter)

  Indicates limiting.

- Hard Clipping(Limiter)

  Enables/disables the “Hard clipping” mode.

- Threshold(Limiter)

  Sets the threshold level above which limiting begins.

- Release Time(Limiter)

  Sets the time it takes to release gain reduction.

- Compress(Compressor)

  Indicates compression.

- Ratio(Compressor)

  Sets compression ratio.

- Threshold(Compressor)

  Sets the threshold level above which compression begins.

- Release Time(Compressor)

  Sets the time it takes to release gain reduction.

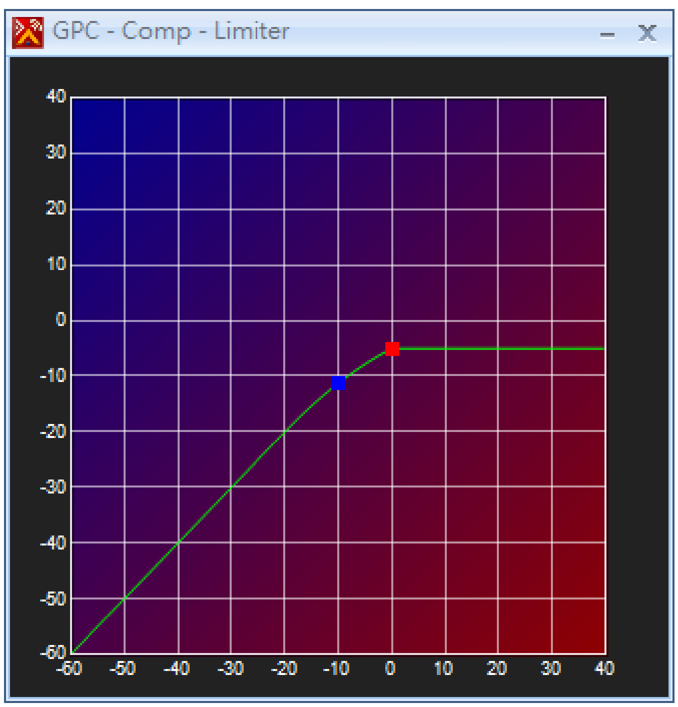
- Attack Time(Compressor)

  Sets the time it takes to respond to the input signal.

- Gain Reduction

  Reflects the current amount of gain reduction.

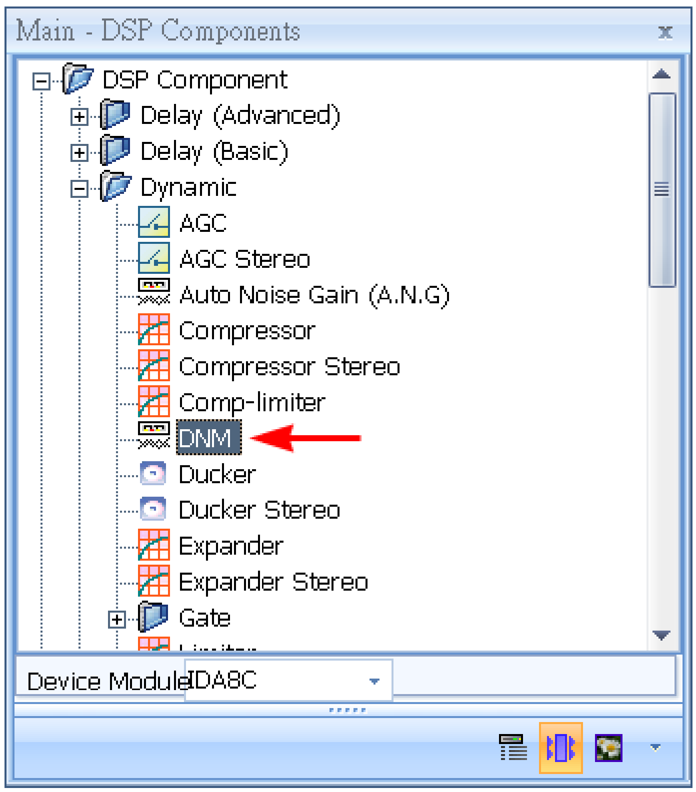
❖ Graphical Control Window



- Axis
  - X: Input signal level (dB)
  - Y: Output signal level (dB)
- Control Points
  - Blue: Adjust Threshold(Compressor).
  - Red: Adjust Threshold(Limiter).

6.4.5 DNM

❖Component Template

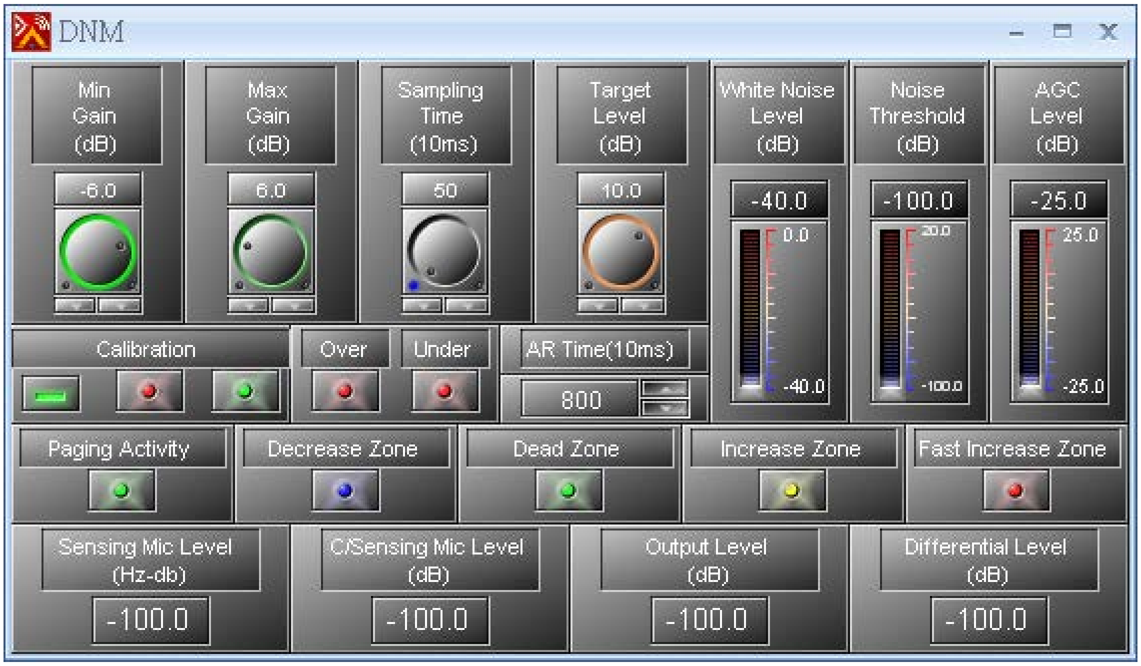


❖Component Appearance

❖Description

DNM component is basically the same as A.N.G component but it use DNM microphone to be the S.M input of A.N.G.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Min Gain	-6.0	-15.0	0	0.1	dB
Max Gain	6.0	0	15.0	0.1	dB
Sampling Time	50	50	60000	50	10ms
Target Level	10.0	15.0	15.0	0.1	dB
White Noise Level	-40.0	-40.0	0	0.1	dB
Noise Threshold	-100.0	-100.0	20.0	0.1	dB
AGC Level	-15.0	-15.0	15.0	0.1	dB
Calibration	Off	-	-	-	-
Calibration Proceeding	Off	-	-	-	-
Calibration Finish	Off	-	-	-	-
Over	Off	-	-	-	-
Under	Off	-	-	-	-
AR Time	8000	500	10000	10	ms
Paging Activity	Off	-	-	-	-
Decrease Zone	Off	-	-	-	-
Dead Zone	Off	-	-	-	-
Increase Zone	Off	-	-	-	-
Fast Increase Zone	Off	-	-	-	-
Sensing Mic Level	-100.0	-100	200	0.1	Hz-dB
C/Sensing Mic Level	-100.0	-100	200	0.1	dB
Output Level	-100.0	-100	200	0.1	dB
Differential Level	-100.0	-100	200	0.1	dB

❖ Element Description

- Min Gain

  The minimum AGC level compensation.

- Max Gain

  The maximum AGC level compensation.

- Sampling Time

  Determine the period time between two ambient noise measures. The less value is more sensitivity, it means the compensation gain of paging input is response more quickly of environment noise if the sampling time is smaller.

- Target Level

  Determine which relative volume you want to target.

- White Noise Level

  White noise level, automatically Adjusted by the calibration.

- Noise Threshold

  The level of background noise measured when calibration button pressed.

- AGC Level

  Display the current compensation gain.

- Calibration

  Click on to start the automatic calibration procedure.

- Calibration Proceeding

  Lit all along the calibration procedure.

- Calibration Finish

  Green led, light on if the calibration procedure is success.

- Over

  Indicates the calibration result:

  The white noise level needed is over the range. A pop-up window will shows you what to do.

- Under

  Indicates the calibration result:

  The white noise level needed is under the range.A pop-up window will show you what to do.

- AR Time

  Attack and release time.

- Paging Activity

  Light up when the paging is active.

  (when P.M. is above the mode threshold).

- Decrease Zone

  Indicates that the AGC gain is decreasing.

- Dead Zone

  Indicates that the AGC is not changing.

- Increase Zone

  Indicates that the AGC gain is increasing.

- Fast Increase Zone

  Indicates that the AGC gain is increasing fast.

- Sensing Mic Level

  Show the input level of the sensing microphone.

- C/Sensing Mic Level

  Show the input level of the sensing microphone through the internal calibration transfer function.

- Output Level

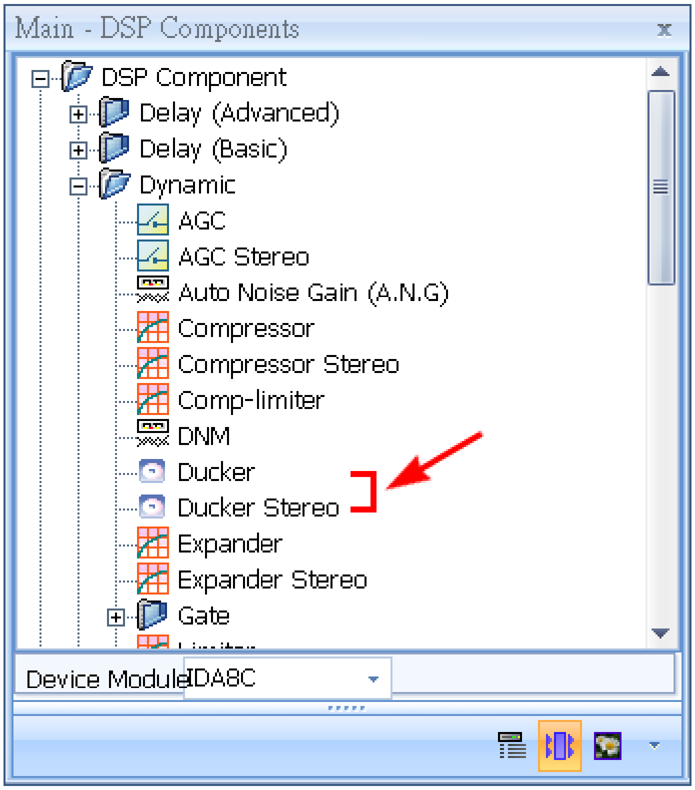
  Show the output level.

- Differential Level

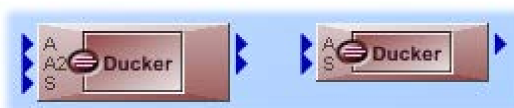
  Difference between the Output level and C/Sensing Mic level. (todo: check with kevin,)

6.4.6 Ducker\Ducker Stereo

❖Component Template



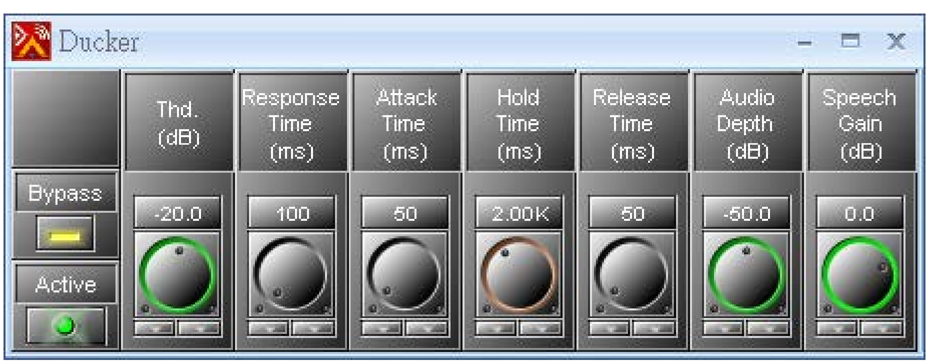
❖Component Appearance



❖Description

The Ducker is a kind of “sophisticated switch” allowing to attenuate or switch off automatically the signal arriving in input A when a chosen level (in dB) is perceived in input S.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Active	Off	-	-	-	-
Thd.	-20.0	-80.0	35.0	0.1	dB
Response Time	100	10	5000	10	ms
Attack Time	50	10	1000	10	ms
Hold Time	2000	10	5000	10	ms
Release Time	50	10	1000	10	ms
Audio Depth	-50	-100	0	0.1	dB
Speech Gain	0	-60.0	20.0	0.1	dB

❖ Element Description

- Bypass

  Switches off the Ducker's operations.

- Active

  Light on when the Ducker is active (Signal has been detected on input S).

- Thd.

  Threshold of detection on the input channel (S). The input channel (A) is attenuate when input channel (S) goes above this threshold.

  Advice : To avoid unexpected attenuation, don't set a low threshold level .

- Response Time

  Adjust the time between the S level's detection and the beginning of the Ducker’s operation on the A signal.

  Advice : Don't set a long "response time", otherwise you won't hear the beginning the (S) signal (as the first words of the speech).

- Attack Time

  Adjust the fade-in time of the (A) signal when Ducker’s is working.

- Hold Time

  Adjust the time between the end (S) level's detection and the end of the Ducker’s operation.

  Advice : Set enough "Hold time" to avoid unexpected background level during the speech's breaks.

- Release Time

  Adjust the fade-out time of the A signal when Ducker’s is no longer working.

- Audio Depth

  Adjusts the A level when the Ducker is active.

- Speech Gain

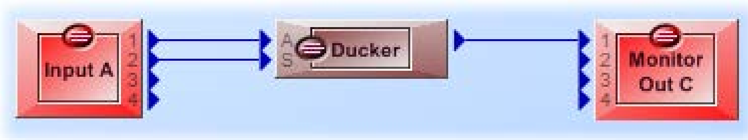
  Adjusts the S level when the Ducker is active.

❖ Use the Ducker Component

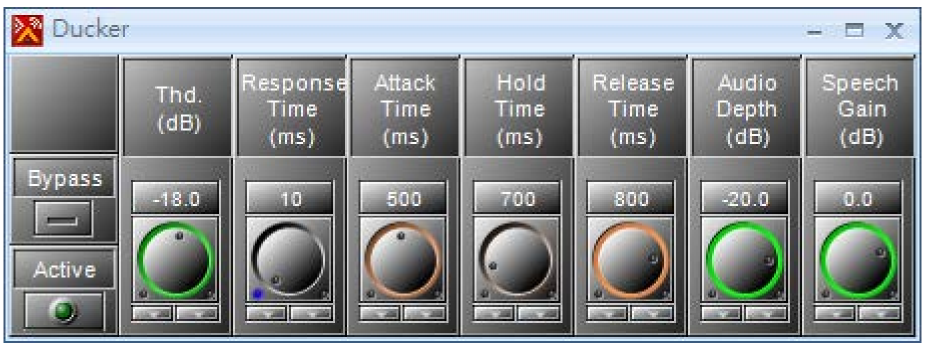
As simple example, we will set a system for a speaker with background music. When the speaker is talking, the background music must decrease its level.

The speaking is using a standard dynamic microphone type SM58. The microphone is connected to the channel 2 of the "Input A" component, with a input sensitivity at -40dB (set in the input A component). Then the microphone is linked to the input (S) of the Ducker.

The music player (as a CD player) is connected to the input 1 of the "Input A" component. Then the music is linked to the input (A) of the Ducker.



Elements:



- Threshold

  Set at -18dB, because less would active the attenuation when the speaker is not talking if the microphone catch some unexpected noise.

- Response Time

  Set very short, because the background music must be attenuated as soon as the speaker is talking.

- Attack and Release Times

  Which are the "fade in" and "fade out" on the background music. Set almost one second for the hearing comfort, we want a kind music transition.

- Hold Time

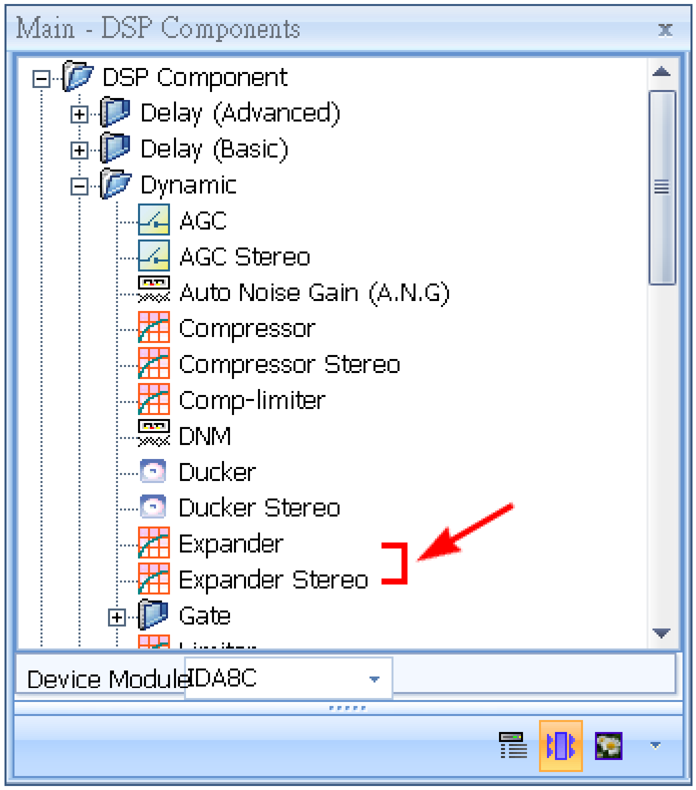
  Set at 700(ms), enough to avoid the music back during the speech breaks.

- Audio Depth

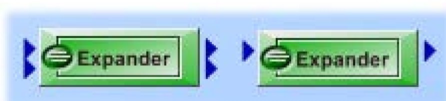
  Set at -20dB, to hear the music during the speech, but at very low level (depending of the context).

6.4.7 Expander\Expander Stereo

❖Component Template



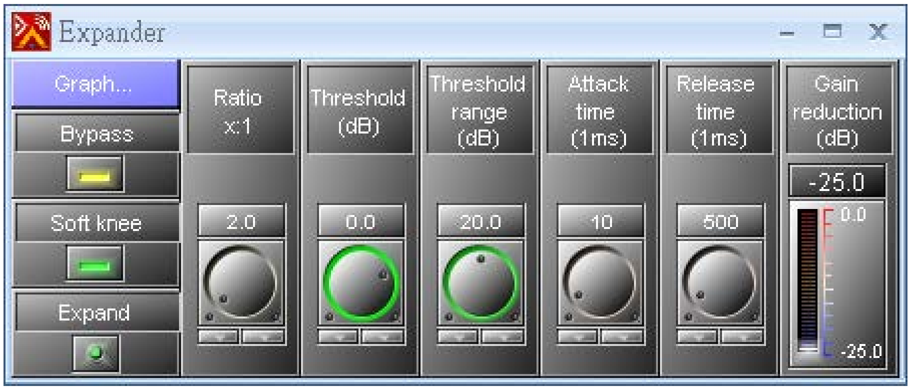
❖Component Appearance



❖Description

An expander works in an opposite fashion than the compressor, the threshold is set and any part of the signal dropping below this threshold will be affected by the expander and this level will be raised. The expander therefore like the compressor balances out the signal making it sound more professional. The expander able to have a signal with a bigger dynamic range.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Soft Knee	On	-	-	-	-
Expand	Off	-	-	-	-
Ratio	2.0	1.0	20.0	0.1	x;1
Threshold	0	-60.0	20.0	0.1	dB
Threshold Range	20.0	0	40.0	0.1	dB
Attack Time	10	1	100	1	ms
Release Time	500	100	5000	1	ms
Gain Reduction	-25.0	-25.0	0	0.1	dB

❖Element Description

- Bypass

  Bypasses the expander's function.

- Soft Knee

  Enables/disables the “Soft Knee” mode.

- Expand

  Indicates that the function is active.

- Ratio

  Sets expansion ratio.

- Threshold

  Sets the threshold level above which expansion begins.

- Threshold Range

  Sets the range of level, in which the ratio gradually alters from 1:1 to the value set by the ratio parameter (soft knee function).

- Attack Time

  Sets the time it takes to respond to the input signal.

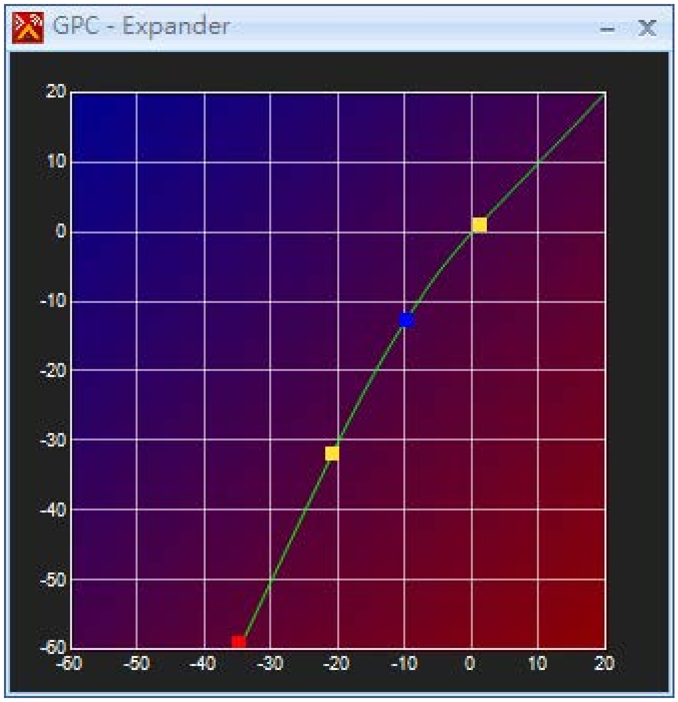
- Release Time

  Adjusts the time to stop the gain reduction.

- Gain Reduction

  Reflects the current amount of gain reduction.

❖Graphical Control Window



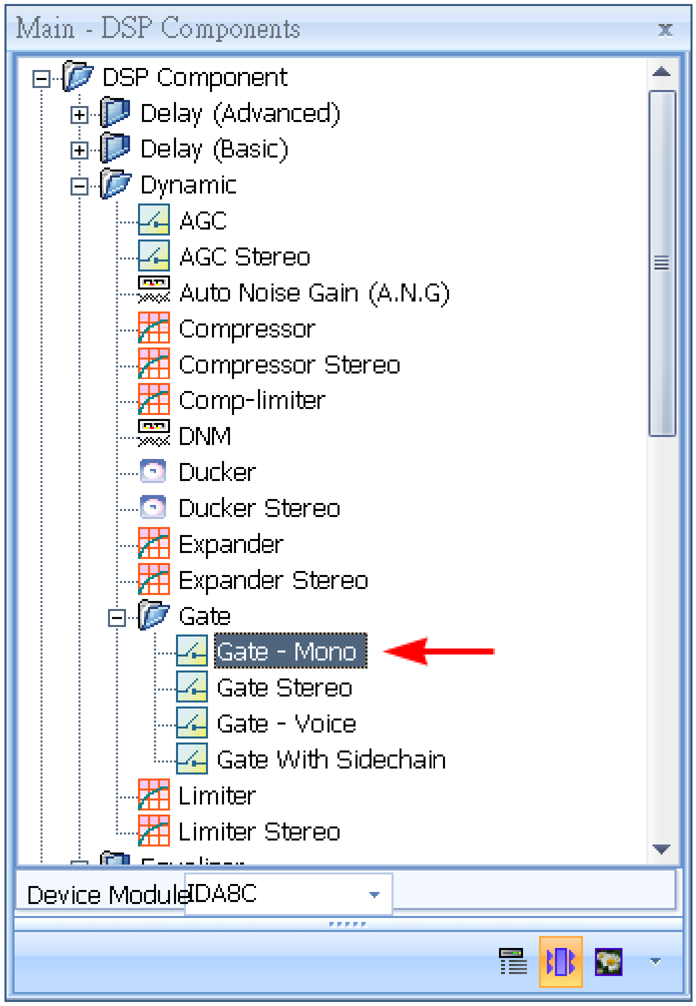
- Axis
  - X: Input signal level(dB)
  - Y: Output signal level(dB)
- Control Points
  - Red: Adjust Ratio.
  - Yellow: Adjust Threshold Range.
  - Blue: Adjust Threshold.

❖

6.4.8 Gate

6.4.8.1 Gate - Mono

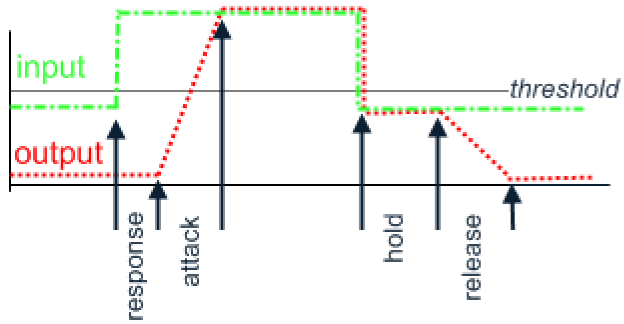
❖ Component Template



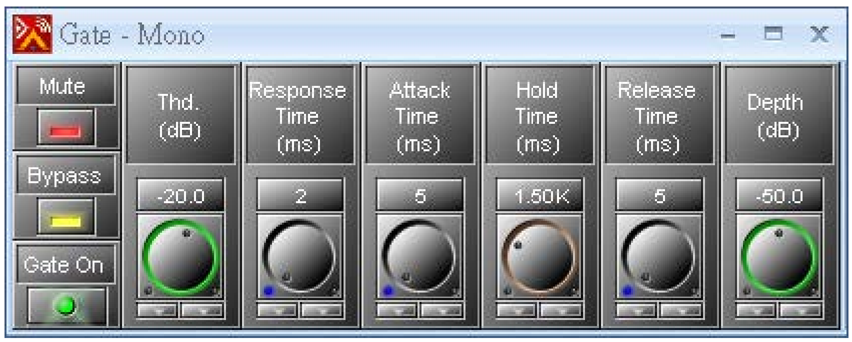
❖ Component Appearance

❖ Description

The gate allows you to cut off the signal below a chosen threshold.



❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	Off	-	-	-	-
Bypass	Off	-	-	-	-
Gate On	Off	-	-	-	-
Thd.	-20.0	-90.0	35.0	0.1	dB
Response Time	2	2	5000	1	ms
Attack Time	5	1	1000	1	ms
Hold Time	1500	1	5000	1	ms
Release Time	5	1	1000	1	ms
Depth	-50.0	-100.0	0	0.1	dB

❖Element Description

- Mute

  Mute the gate's module.

- Bypass

  Switch off the gate's operation.

- Gate On

  Light on when the gate is active (open).

- Thd.

  Sets the threshold level above which the Gate will open.

- Response Time

Adjust the time between the level's detection and the beginning of the Gate operation.

- Attack Time

  Adjust the fade-in time of the signal when Gate is working.

- Hold Time

  Adjust the time between the end level's detection and the end of the Gate operation.

- Release Time

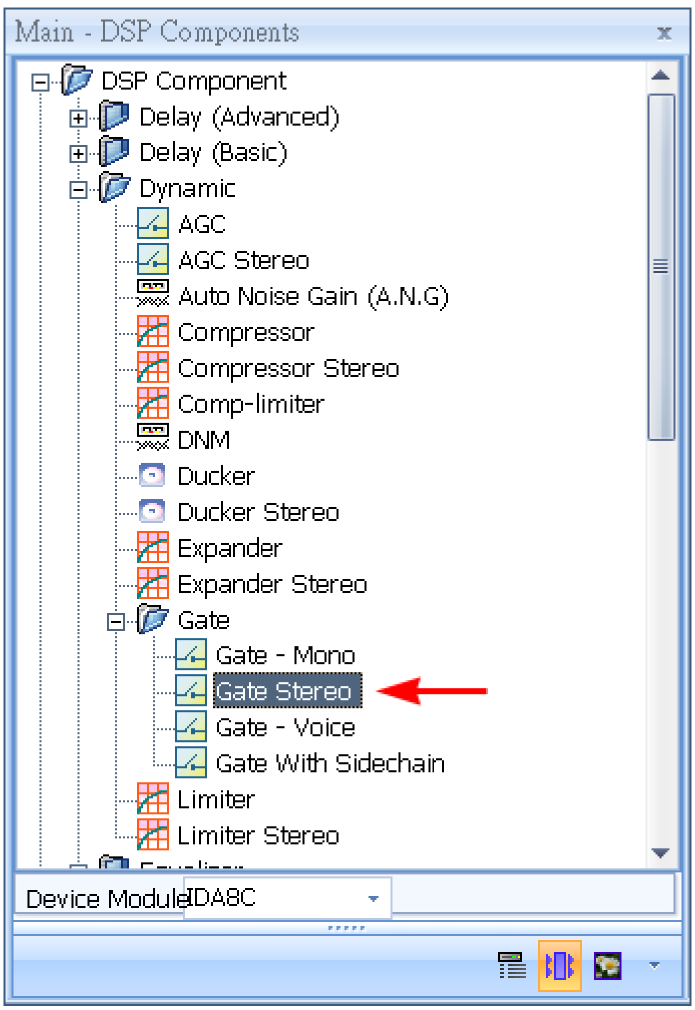
  Adjust the fade-out time of the signal when Gate is no longer working.

- Depth

  Attenuate the level when the Gate is active.

6.4.8.2 Gate - Stereo

❖ Component Template

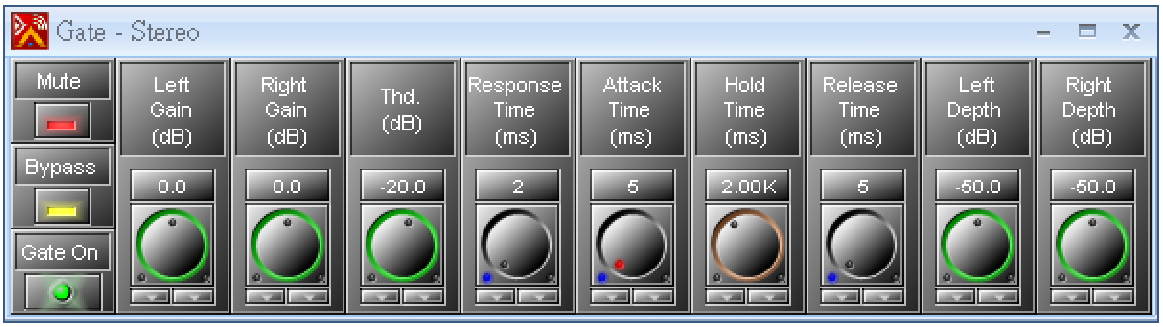


❖ Component Appearance

❖ Description

Simple double channel Gate.

❖ Control Window



❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	Off	-	-	-	-
Bypass	Off	-	-	-	-
Gate On	Off	-	-	-	-
Left Gain	0	-35.0	35.0	0.1	dB
Right Gain	0	-35.0	35.0	0.1	dB
Thd.	-20.0	-90.0	35.0	0.1	dB
Response Time	5	1	1000	1	ms
Attack Time	2	1	5000	1	ms
Hold Time	2000	1	5000	1	ms
Release Time	5	1	1000	1	ms
Left Depth	-50.0	-100.0	0	0.1	dB
Right Depth	-50.0	-100.0	0	0.1	dB

❖ Element Description

- Mute

  Mute the gate's module.

- Bypass

  Switch off the gate's operation.

- Gate On

  Light on when the gate is active (open).

- Left Gain

  Set the volume of the signal connected in the input L.

- Right Gain

Set the volume of the signal connected in the input R.

- Thd.

  Sets the threshold level above which Gate will open.

- Response Time

  Adjust the time between the level's detection and the beginning of the Gate operation.

- Attack Time

  Adjust the fade-in time of the signal when Gate is working.

- Hold Time

  Adjust the time between the end level's detection and the end of the Gate operation.

- Release Time

  Adjust the fade-out time of the signal when Gate is no longer working.

- Left Depth

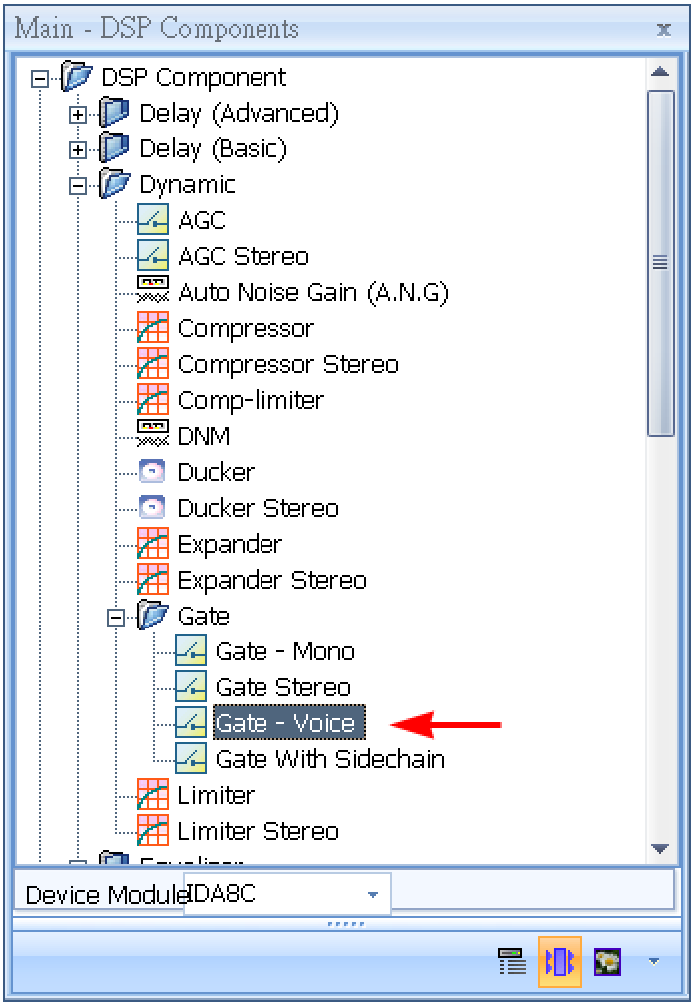
  Attenuate the level when the Gate is active.

- Right Depth

  Attenuate the level when the Gate is active.

6.4.8.3 Gate - Voice

❖ Component Template



❖ Component Appearance

❖ Description

This component is gate which is using an active detection algorithm specially dedicated for voice. The gate is open only if a human voice pass is detected.For example you knock table, this noise will be stopped by Gate-Voice, but when you speak then the voice gate will allow the human voice to open the gate. If there is a noise and a human voice present, the voice gate will open and both noise and voice will get through.

⚠ This is NOT a noise cancellation component.

❖ Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Gate On	Off	-	-	-	-
Mute	Off	-	-	-	-
Bypass	Off	-	-	-	-
Thd.	-20.0	-90.0	35.0	0.1	dB
Hold Time	2000	10	5000	10	ms
Release Time	50	10	1000	10	ms
Voice Recognition	Soft	-	-	-	-

Item list of Voice Recognition:

No.	Name
1	Soft
2	Medium
3	Accurate

❖Element Description

- Gate On

  Light on when the Gate-Voice is active (open).

- Mute

  Mute the Gate-Voice's module.

- Bypass

  Switch off the Gate-Voice's operation.

- Thd.

  The threshold to determine to pass input signal.

- Hold Time

  Time to close the output when no more human speech is detected.

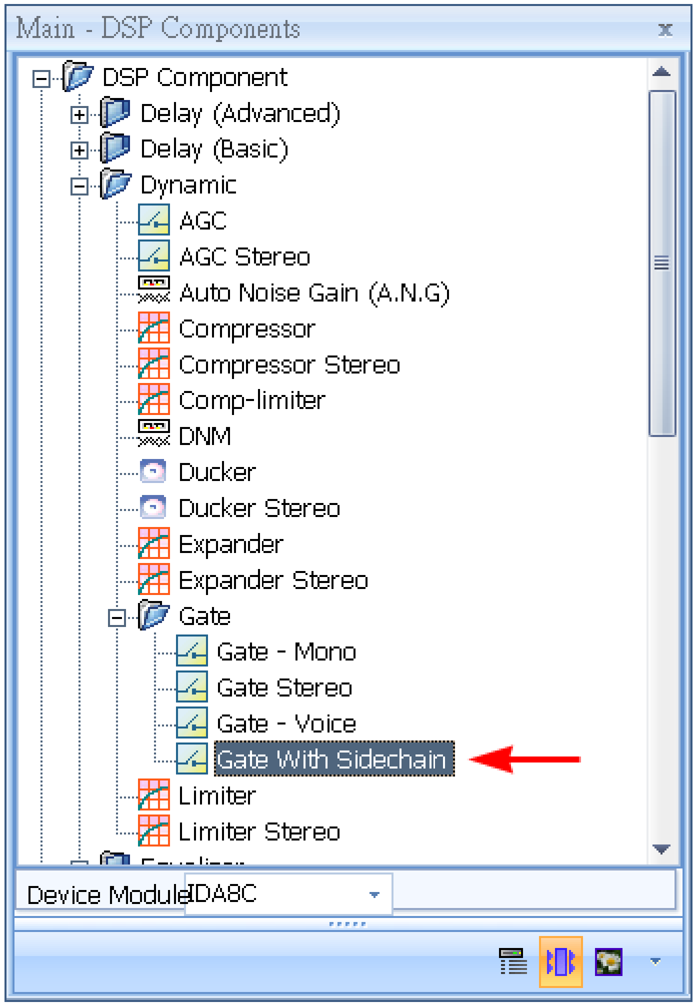
- Release Time

  Adjust the fade-out time of the signal to decrease the signal after the hold time.

- Voice Recognition

6.4.8.4 Gate with Sidechain

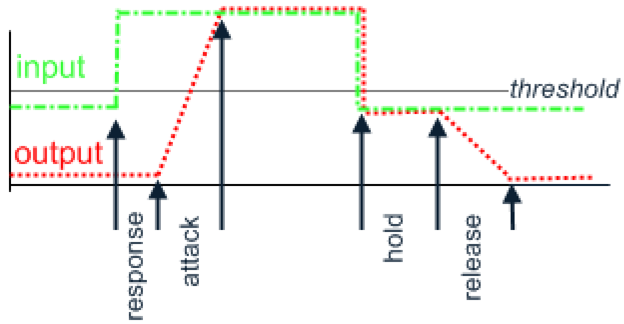
❖ Component Template



❖ Component Appearance

❖ Description

With this gate, you can control a gate on an audio channel (I) with an external Sidechain signal(S). When the Sidechain input (S) input goes above the threshold, the gate becomes active and the signal on Input (I) goes to the output.



❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	Off	-	-	-	-
Bypass	Off	-	-	-	-
Gate On	Off	-	-	-	-
Sidechain Gain	0	-35.0	35.0	0.1	dB
Input Gain	0	-35.0	35.0	0.1	dB
Thd.	-20	-90.0	35.0	0l.1	dB
Response Time	2	1	5000	1	ms
Attack Time	5	1	1000	1	ms
Hold Time	2000	1	5000	1	ms
Release Time	5	1	1000	1	ms
Depth	-50.0	-100.0	0	0.1	dB

❖Element Description

- Mute

  Mute the gate's module.

- Bypass

  Switch off the gate's operation.

- Gate On

Flash when the gate is active (open).

- Sidechain Gain

  Set the volume of the signal connected in the input S.

- Input Gain

  Set the volume of the signal connected in the input I.

- Thd.

  Sets the threshold level (input S) above which Gate will open.

- Response Time

  Adjust the time between the level's detection and the beginning of the Gate operation.

- Attack Time

  Adjust the fade-in time of the signal when Gate is working.

- Hold Time

  Adjust the time between the end level's detection and the end of the Gate operation.

- Release Time

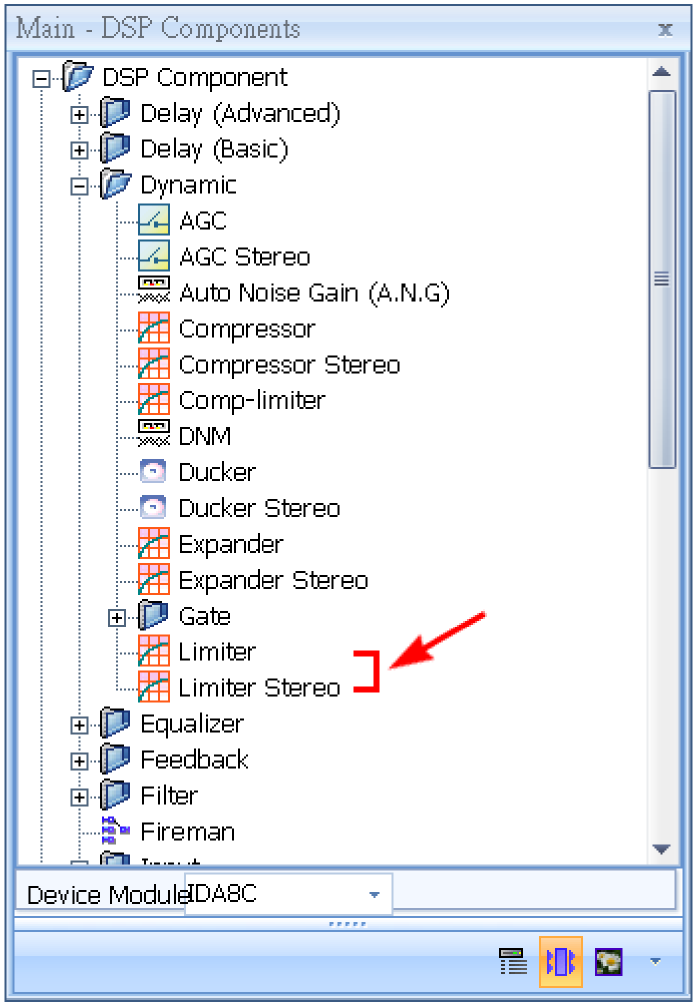
  Adjust the fade-out time of the signal when Gate is no longer working.

- Depth

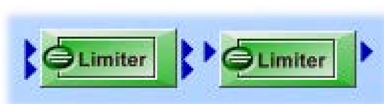
  Attenuate the level when the Gate is active.

6.4.9 Limiter\Limiter Stereo

❖Component Template



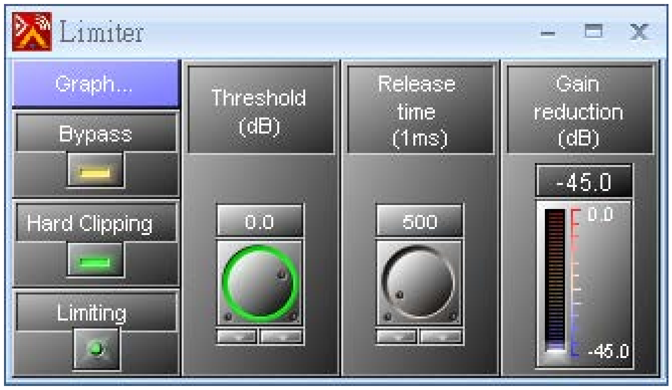
❖Component Appearance



❖Description

This component limits the audio level depending on a threshold.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Hard Clipping	On	-	-	-	-
Limiting	Off	-	-	-	-
Threshold	0	-60.0	20.0	0.1	dB
Release Time	500	100	5000	1	ms
Gain Reduction	-45.0	-45.0	0	0.1	dB

❖Element Description

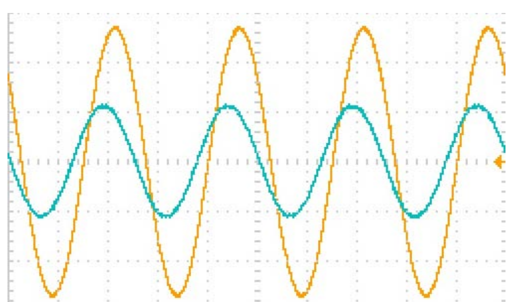
- Bypass

  Switch off the Limiter's operation

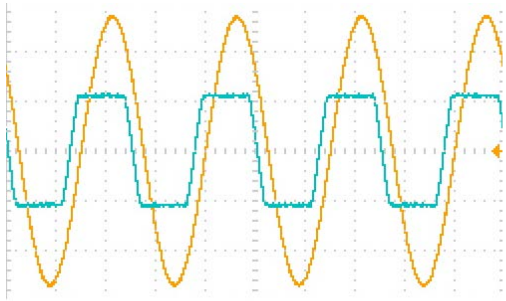
- Hard Clipping

  Enables/disables the “Hard clipping” mode.

  This is an example of a sinusoidal signal limited with and without Hard Clipping.



Without Hard Clipping



With Hard Clipping

- Limiting

  Indicates limiting.

- Threshold

  Sets the threshold level above which limiting begins.

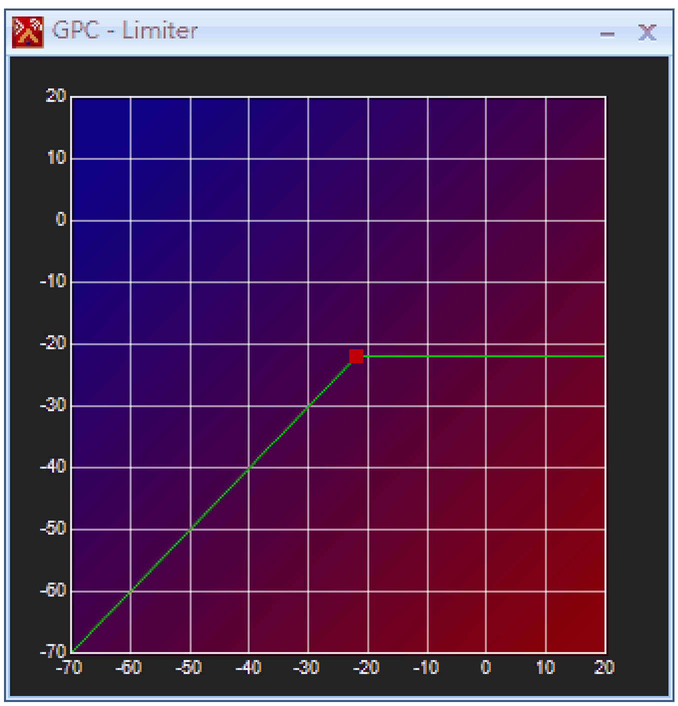
- Release Time

  Sets the time it takes to release gain reduction.

- Gain Reduction

  Reflects the current amount of gain reduction.

❖ Graphical Control Window



- Axis
  - X: Input signal level(dB)

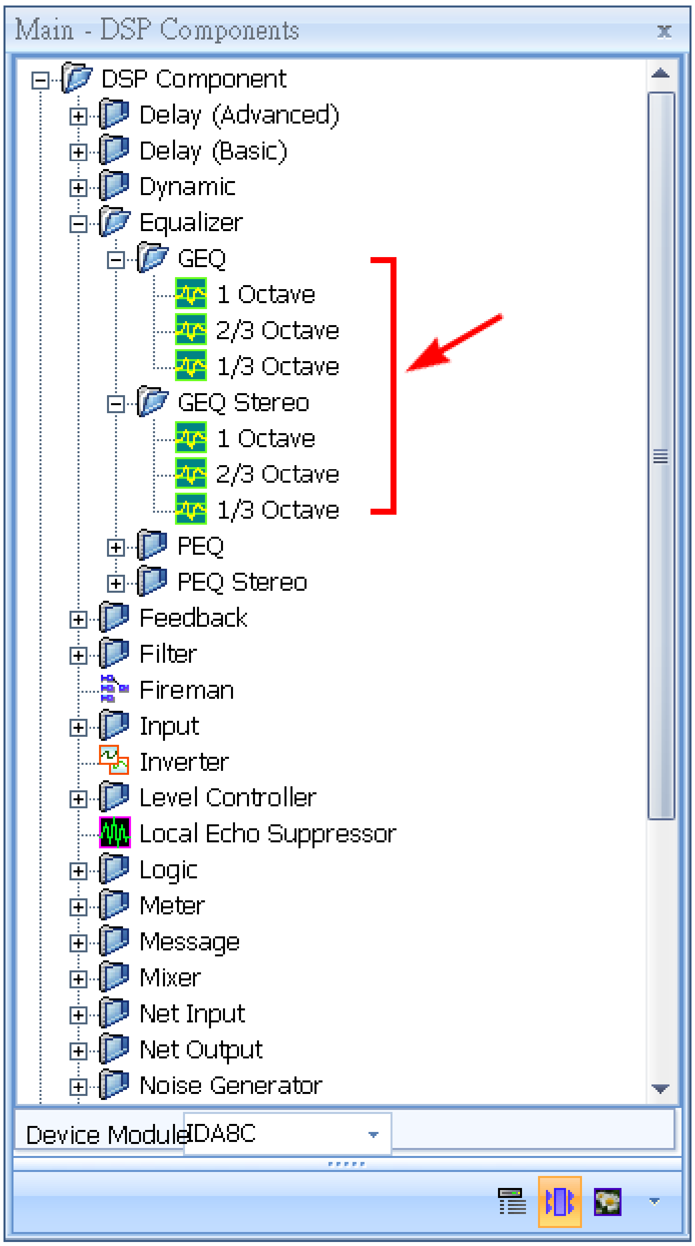
- Y: Output signal level(dB)

- Control Points
  - Red: Adjust Threshold.

6.5 Equalizer

6.5.1 GEQ\GEQ Stereo

❖Component Template



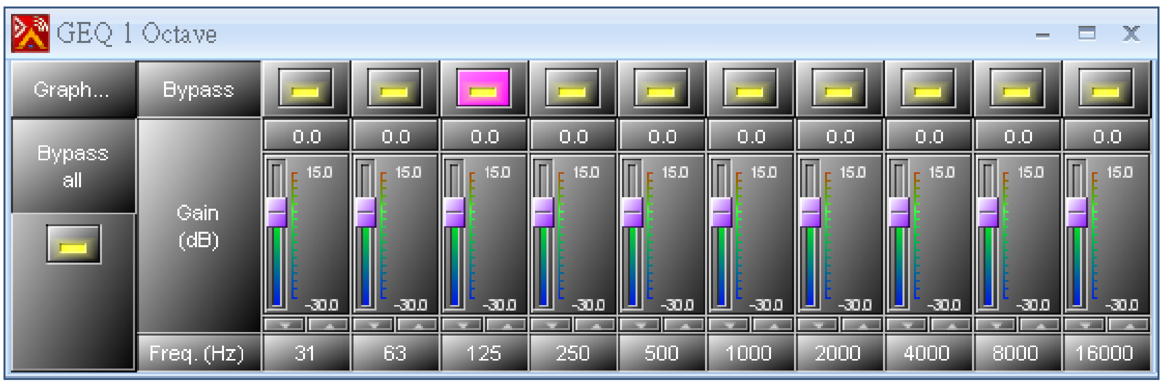
❖Component Appearance

Description

GEQ is abbreviation of graphic equalizer, in the graphic equalizer, the input signal is sent to a bank of filters. Each filter passes the portion of the signal present in its own frequency range or band. The amplitude passed by each filter is adjust using a slider in control window of Ateis Studio control to boost or cut frequency components passed by that filter. The vertical position of each slider thus indicates the gain applied at that frequency band.

The number of frequency channels (and therefore each one's bandwidth) affects the consuming DSP power of audio processor, and may be matched to the requirements of the intended application. An equalizer for professional live sound reinforcement typically has some 25 to 31 bands, for more precise control of feedback problems and equalization of room modes. There are three types of GEQ, 1 Octave, 2/3 Octave and 1/3 Octave. 1/3 Octave GEQ (spoken informally as "third-octave EQ") because the center frequency of its filters are spaced one third of an octave apart, three filters to an octave. Equalizers with half as many filters per octave are common where less precise control is called a 2/3-octave equalizer, and so on the 1 Octave GEQ, that is one filters to an octave.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass all	Off	-	-	-	-
Bypass	Off	-	-	-	-
Gain	0	-30.0	15.0	0.1	dB

❖Element Description

- Bypass all

  To bypass the EQ (all the frequency band).

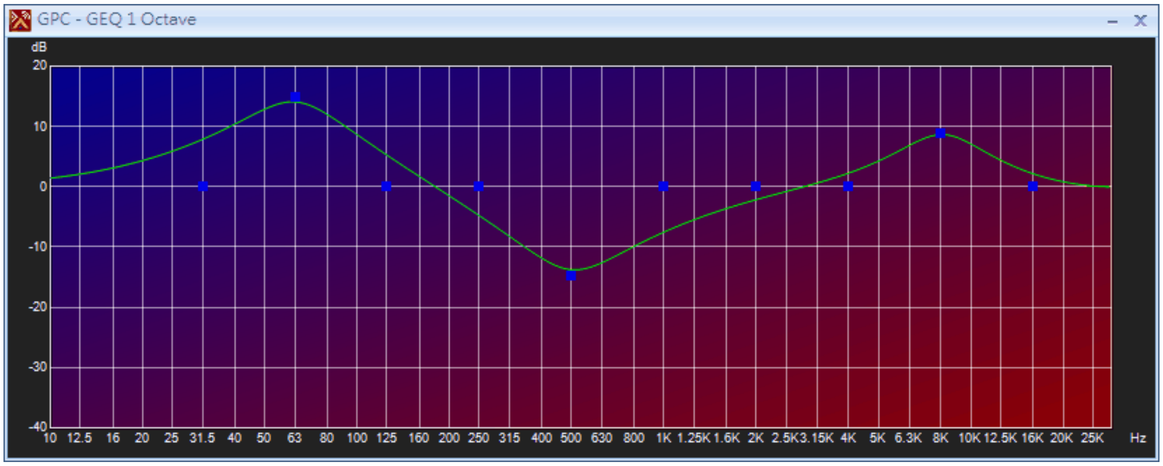
- Bypass

  Bypasses only the selected frequency band.

- Gain

  Set the gain of the selected frequency band.

❖Graphical Control Window



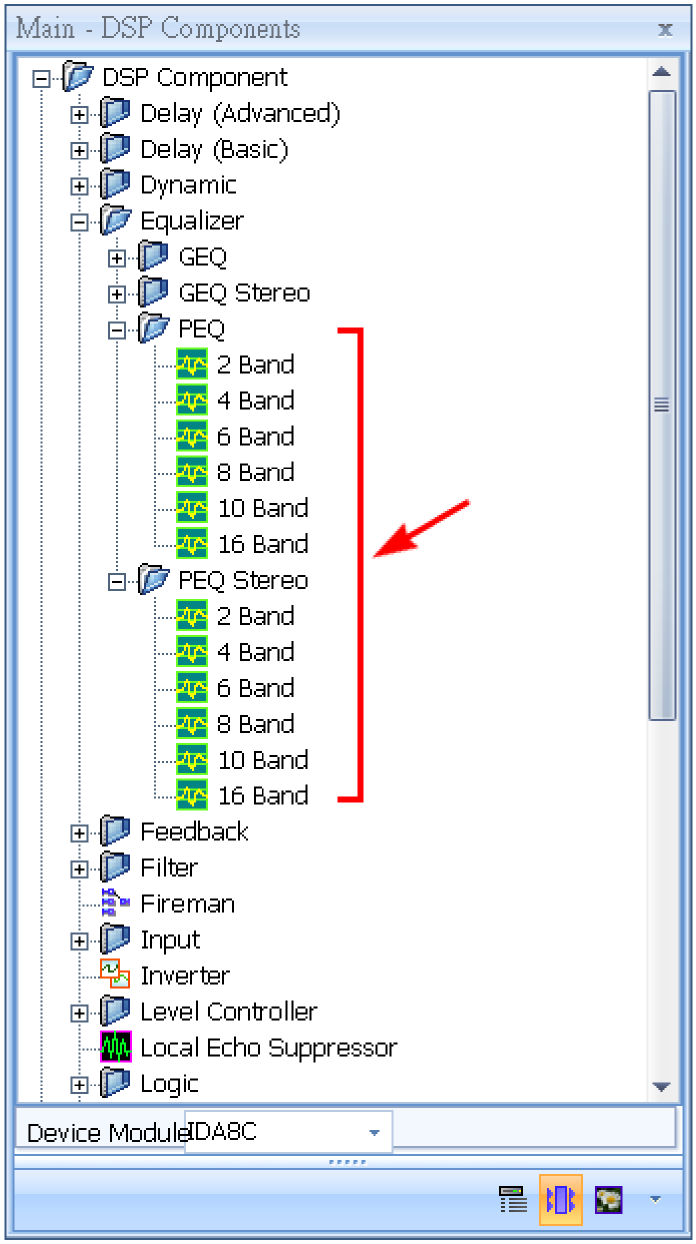
- Axis
  - X: Frequency of output signal(Hz)
  - Y: Gain of output signal(dB)
- Control Points
  - Blue: Adjust Gain.

❖ Application

Equalizers can correct problems posed by a room's acoustics, as an auditorium will generally have an uneven frequency response especially due to standing waves and acoustic dampening. The frequency response of a room may be analyzed using a spectrum analyzer and a pink noise generator for instance. Then a graphic equalizer can be easily adjusted to compensate for the room's acoustics. Such compensation can also be applied to tweak the sound quality of a recording studio in addition to its use in live sound reinforcement systems.

6.5.2 PEQ\PEQ Stereo

❖Component Template



❖Component Appearance

Description

PEQ is abbreviation of parametric equalizer, PEQs are multi-band variable equalizers which allow users to control the three primary parameters: amplitude, center frequency and bandwidth. The amplitude of each band can be controlled, and the center frequency can be shifted, and bandwidth ("Q") can be widened or narrowed. PEQs are capable of making much more precise adjustments to sound than other equalizers, and are commonly used in sound recording and live sound reinforcement.

❖Control Window

❖Element Properties

- Generate Information

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass all	Off	-	-	-	-
Bypass	Off	-	-	-	-
Bandwidth	1.00	1.00	4.00	0.01	Octive
Frequency	*1	10	22000	1	Hz
Gain	0	-30.0	15.0	0.1	dB

*1: The initial value of Frequency depends on how many bands of a PEQ. These initial values are average distribution in range 10 to 22000 Hz.

❖Element Description

- Bypass all

  To bypass the EQ (all the frequency band).

- Bypass

  Bypasses only the selected frequency band.

- Bandwidth

  Set the width around the frequency (Q factor) of selected frequency band.

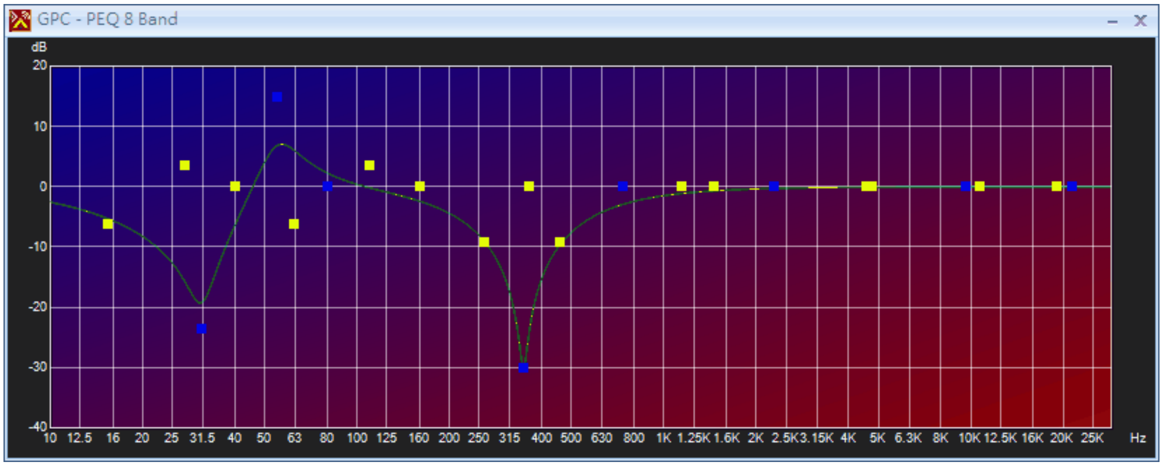
- Frequency

  Set the frequency of a band.

- Gain

Set the gain (attenuate or increase the selected frequency).

❖ Graphical Control Window



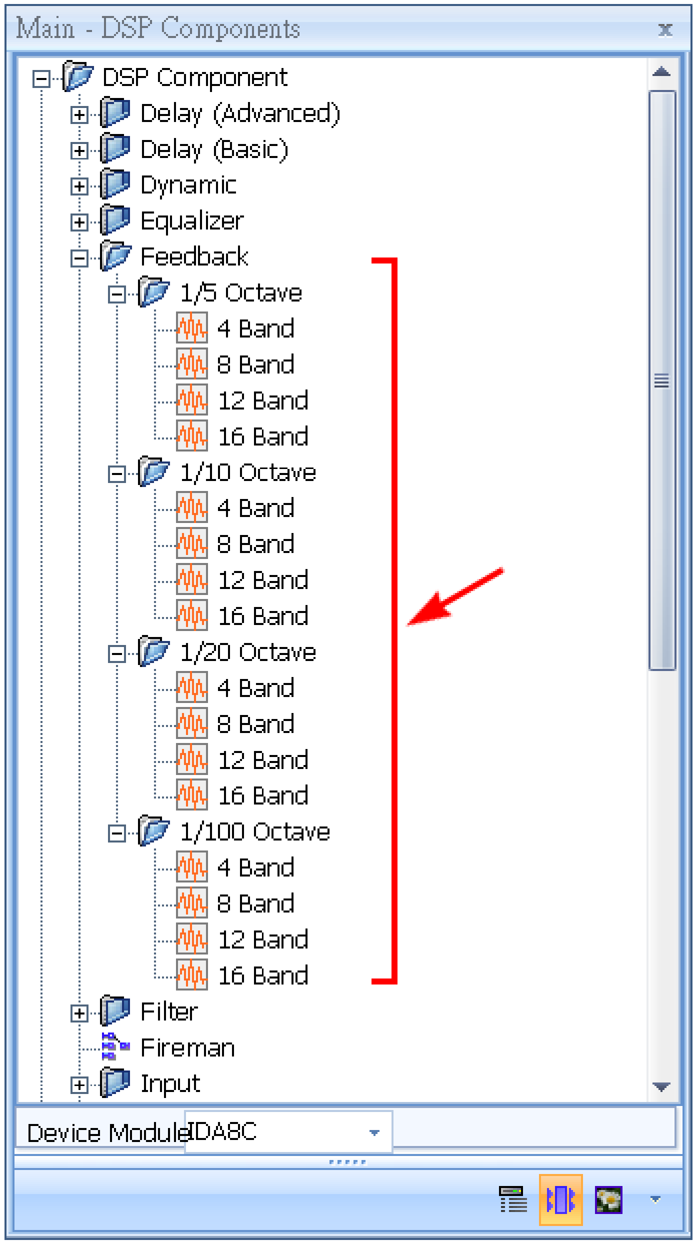
- Axis
  - X: Frequency of output signal(Hz)
  - Y: Gain of output signal(dB)
- Control Points
  - Blue: Adjust Gain(using vertical direction) and Frequency(using horizontal direction).
  - Yellow: Adjust Bandwidth

❖ Application

An equalizer can be used to correct or "flatten" the frequency response of speakers rather than designing the speaker itself to be equalized. For some speaker system doesn't use separate woofers and tweeters to cover the bass and treble frequencies, but includes full-range drivers. However this speaker system is sold with an active equalizer designed to correct the poor frequency balance of those drivers. That equalizer must into the sound system but before amplifier so that the amplified signal that is finally sent to the speakers has its response increased at the frequencies where the response of these drivers falls off, producing a high fidelity reproduction regardless. Ateis audio processor give you the same solution without using an external equalizer, all it done in software component PEQ.

6.6 Feedback

❖Component Template



❖Component Appearance

❖Description

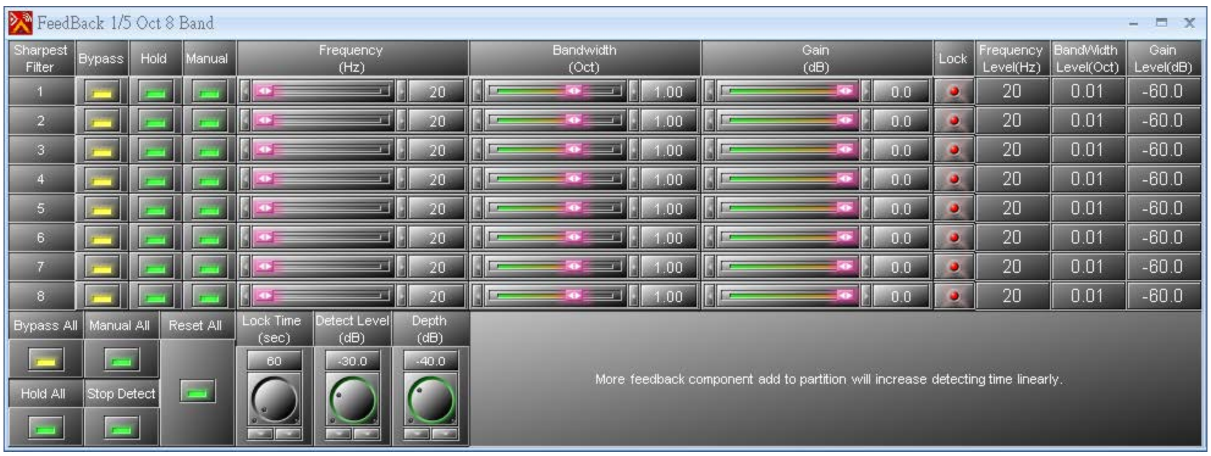
The error between the desired and the actual output is taken and given as feedback to the adaptive processor for adjusting its coefficients to minimize the error. The feedback analyzes the signal,

detects feedback and attenuates the responsible frequency. There is several kind of feedback component. The difference is only the bandwidth which are tuned for Music high than speech (between it the bandwidth is decreasing). Each category has 4 different feedback it is only the number of filter (4, 8 12 or 16) that the feedback will use. The treatment time is increased with the number of filter.

use feedback killers to detect feedback then note the frequency and replace feedback component with an Equalizer. And add a simply 4 notch feedback for security.

By storing again the configuration, you will erase the captured Values (frequencies, bandwidth and gain) !

❖Control Window



❖Element Properties

• Generate Information

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Hold	On	-	-	-	-
Manual	Off	-	-	-	-
Frequency	20	20	20000	1	Hz
Bandwidth	1.00	0.01	1.50	0.01	Octive
Gain	0	-60.0	0	0.1	dB
Lock	Off	-	-	-	-
Frequency Level	20	20	20000	1	Hz
Bandwidth Level	0.01	0.01	1.50	0.01	Octive
Gain Level	-60.0	-60.0	0	0.1	dB
Bypass All	-60.0	-	-	-	-
Manual All	Off	-	-	-	-
Hold All	On	-	-	-	-
Stop Detect	Off	-	-	-	-
Reset All	Off	-	-	-	-
Lock Time	60	30	600	1	Second
Detect Level	-30.0	-40.0	0	0.1	dB
Depth	-40.0	-60.0	0	0.1	dB

❖ Element Description

- Bypass

  Bypass the feedback’s module.

  ⚠ By clicking on Bypass you will erase the captured values.

- Hold

  When a Larsen is detected and a frequency locked the component hold this setting.

- Manual

  To enable the Frequency, Bandwidth and Gain fader to be used manually. Unable the automatic detection.

- Frequency

  Frequency of the notch filter to attenuate the feedback.

- Bandwidth

  Bandwidth of the notch filter to attenuate the feedback.

- Gain

  Attenuation of the locked frequency.

- Lock

  Shows if the feedback killers channel has detected and is erased a frequency responsible of a feedback.

- Frequency Level

  Show the frequency locked.

- Bandwidth Level

  Show the bandwidth of the notch filter locked.

- Gain Level

  Show the attenuation of the frequency locked.

- Bypass All

  Bypass all channels, todo

- Manual All

  Active manual settings (with Frequency, Bandwidth and gain knobs).

- Hold All

  Hold all channels, todo behavior description.

- Stop Detect

  todo

- Reset All

todo

- Lock Time

  Interval time during the filter will be active when a feedback is detected. After that the captured values will be loose.

- Detect Level

  Level of feedback detection.

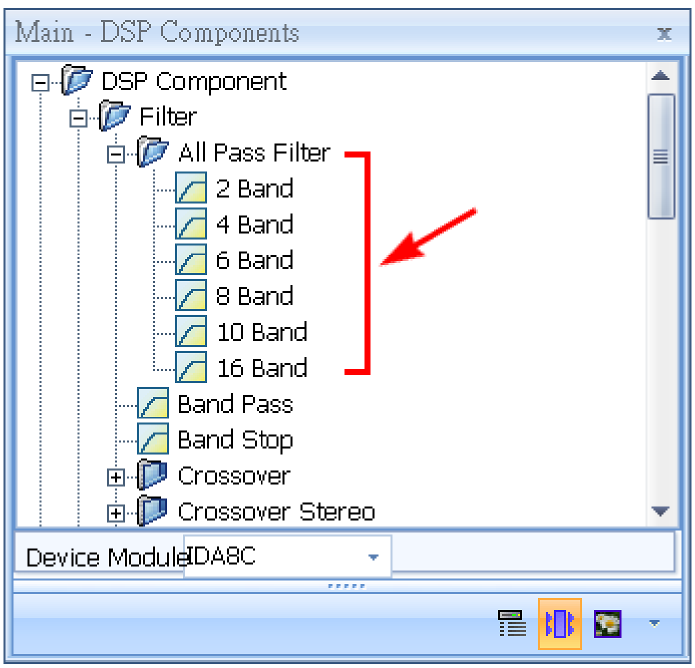
- Depth

  Means maximum reduction gain for filter.

6.7 Filter

6.7.1 All Pass Filter

❖Component Template

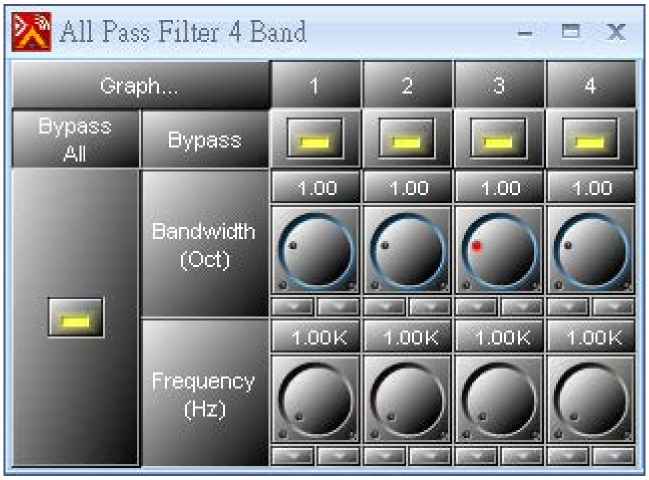


❖Component Appearance

❖Description

An all-pass filter is a signal processing filter that passes all frequencies equally, but changes the phase relationship between various frequencies. It does this by varying its propagation delay with frequency.

❖Control Window



❖Element Properties

- Generate Information

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass All	Off	-	-	-	-
Bypass	On	-	-	-	-
Bandwidth	1.00	0.01	4.00	0.01	Octive
Frequency	1000	20	20000	1	Hz

❖Element Description

- Bypass All

  Bypass the component (all bands of filter's module).

- Bypass

  Bypass the processing of selected band in filter's module.

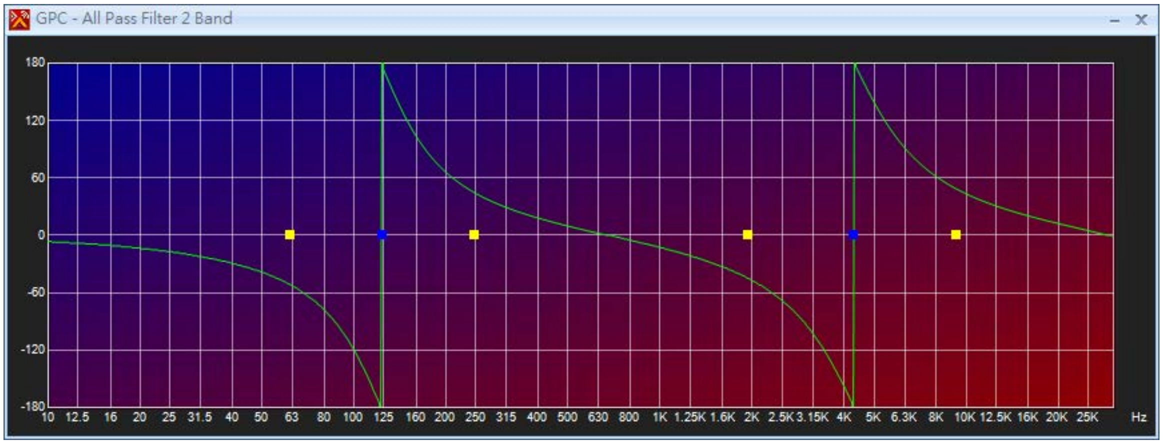
- Bandwidth

  The transition bandwidth.

- Frequency

  The frequency at which the phase shift crosses 90°.

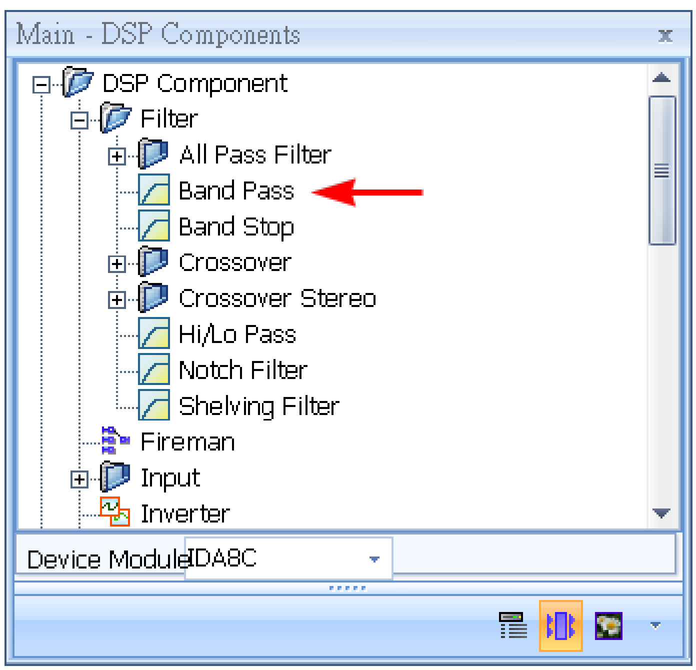
❖Graphical Control Window



- Axis
  - X: Frequency of output signal(Hz)
  - Y: Gain of output signal(dB) (Kevin)
- Control Points
  - Blue: Adjust Frequency
  - Yellow: Adjust Bandwidth

6.7.2 Band Pass

❖Component Template

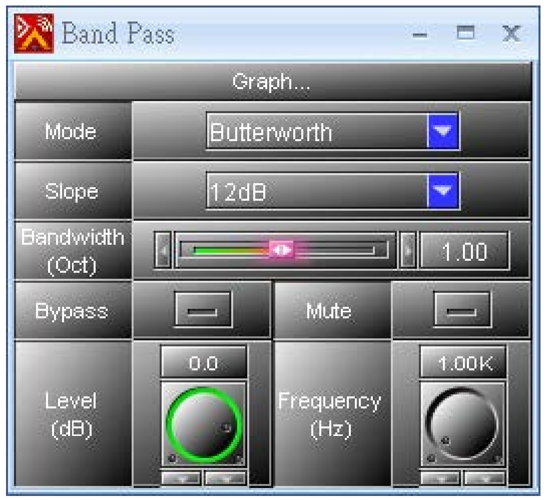


❖Component Appearance

❖Description

A Band Pass component that passes frequencies within a certain range and rejects (attenuates) frequencies outside that range.

❖Control Window



❖Element Properties

- Generate Information

Name	Initial	Minimum	Maximum	Precision	Unit
Mode	Butterworth	-	-	-	-
Slope	12 dB	-	-	-	-
Bandwidth	1.00	0.01	2.00	0.01	Octave
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
Level	0	-90.0	20.0	0.1	dB
Frequency	1000	10	22000	1	Hz

- Item list of Mode:

No.	Name
1	Butterworth

- Item list of Slope:

No.	Name
1	12 dB

❖Element Description

- Mode

Type of filtering,

- Slope

  Slope of attenuation.

- Bandwidth

  Difference between the upper and lower frequencies in a contiguous set of frequencies.(todo Kevin)

- Mute

  Mute the filter's module.

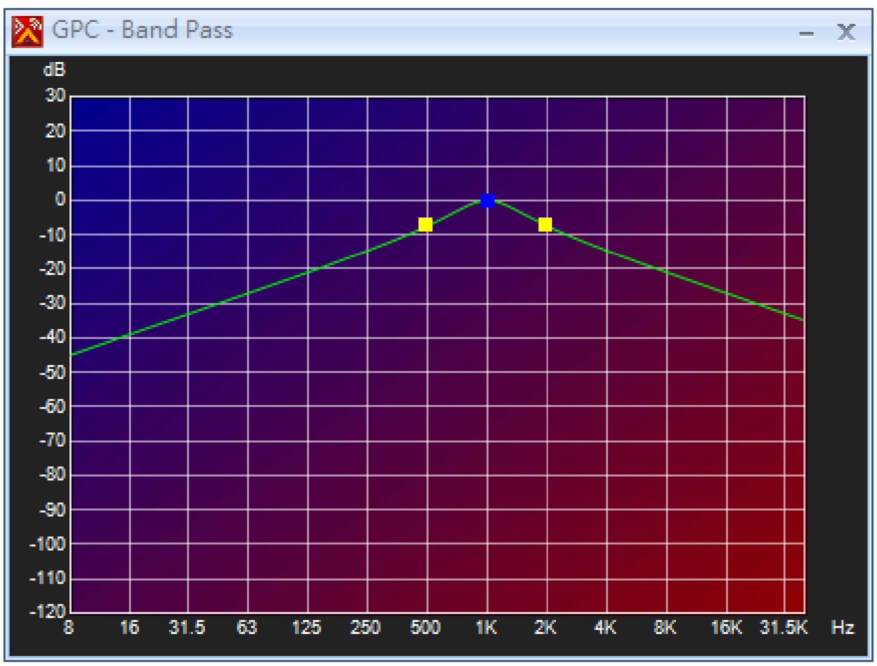
- Level

  Gain of filter's module.

- Frequency

  Frequency of cut.

❖ Graphical Control Window



- Axis

  ○ X: Frequency of output signal(Hz)

  ○ Y: Level of input signal(dB)

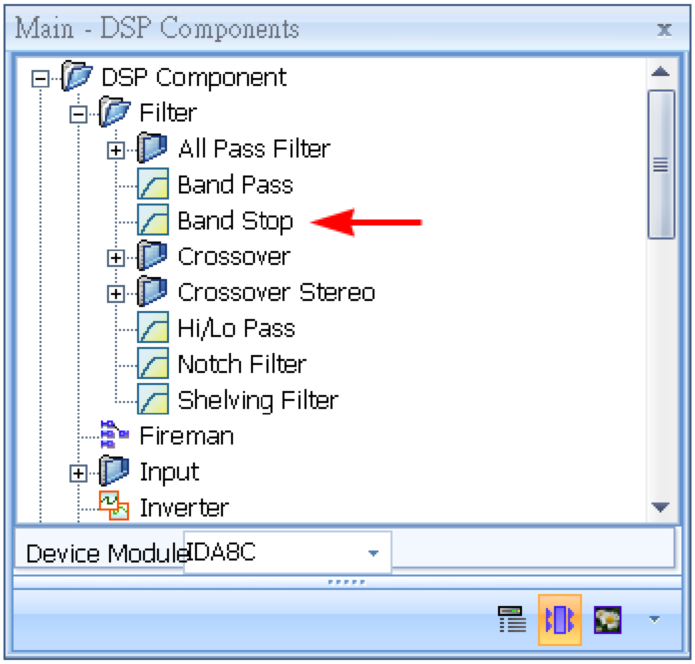
- Control Points

  ○ Blue: Adjust element Frequency(Horizontal) and Level(Vertical).

  ○ Yellow: Adjust element Bandwidth

6.7.3 Band Stop

❖Component Template

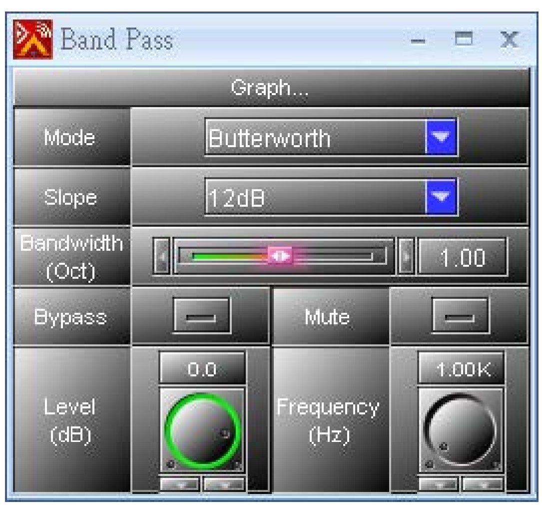


❖Component Appearance

❖Description

A Band Stop filter is a filter that passes most frequencies unaltered, but attenuates those in a specific range to very low levels. It is the opposite of a band-pass filter.

❖Control Window



❖Element Properties

- Generate Information

Name	Initial	Minimum	Maximum	Precision	Unit
Mode	Butterworth	-	-	-	-
Slope	12dB	-	-	-	-
Bandwidth	1.00	0.01	2.00	0.01	Octave
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
Level	0	-90.0	20.0	0.1	dB
Frequency	1000	10	22000	1	Hz

- Item list of Mode:

No.	Name
1	Butterworth

- Item list of Band Stop:

No.	Name
1	12dB

❖ Element Description

- Mode

  Type of filtering.

- Slope

  Slope of attenuation.

- Bandwidth

  Difference between the upper and lower frequencies in a contiguous set of frequencies.(todo Kevin)

- Bypass

  Bypass the filter's module.

- Mute

  Mute the filter's module.

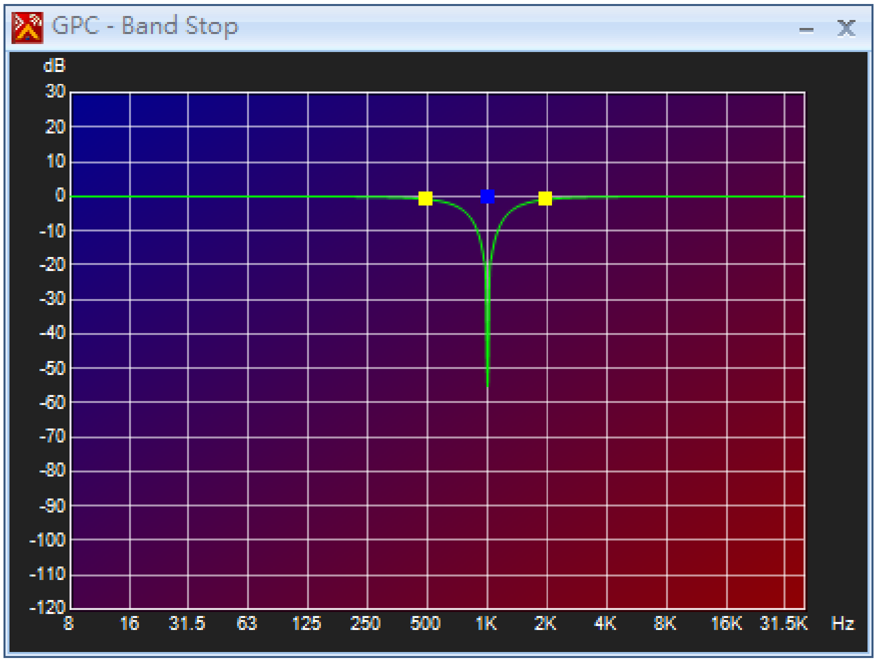
- Level

  Gain of filter's module.

- Frequency

  Frequency of cut.

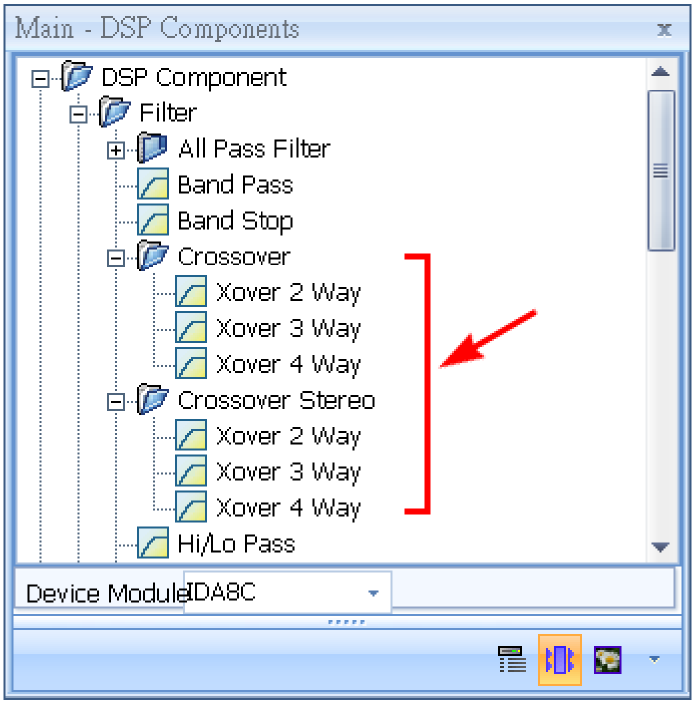
❖ Graphical Control Window



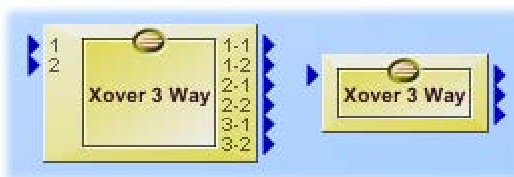
- Axis
  - X: Frequency of output signal(Hz)
  - Y: Level of input signal(dB)
- Control Points
  - Blue: Adjust element Frequency(Horizontal) and Level(Vertical).
  - Yellow: Adjust element Bandwidth

6.7.4 Crossover/Crossover Stereo

❖ Component Template



❖ Component Appearance



❖ Description

Crossovers split the audio signal into separate frequency bands that can be separately routed to other DSP components for those bands. A 2-way crossover consists of a low-pass and a high-pass filter. A 3-way crossover is constructed as a combination of low-pass, band-pass and high-pass filters (LPF, BPF and HPF respectively).

❖ Control Window

todo

❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Invert	-	-	-	-	-
Mode	-	-	-	-	-
Slope	-	-	-	-	-
Mute	-	-	-	-	-
Level	-	-	-	-	-
Frequency	*1	*1	*1	1	Hz

*1: The Initial/Minimum/Maximum values are different between types of Crossover/Crossover Stereo.

Name	Initial	Minimum	Maximum	Precision	Unit
Frequency(2 Way-Low/High Band)	1000	10	22000	1	Hz
Frequency(3 Way-Low Band)	250	20	5590	1	Hz
Frequency(3 Way-Mid Band-Hi Pass)	250	26	20000	1	Hz
Frequency(3 Way-Mid Band-Low Pass)	2500	200	15874	1	Hz
Frequency(3 Way-High Band)	2500	2500	20000	1	Hz
Frequency(4 Way-Low Band)	160	20	634	1	Hz
Frequency(4 Way-Mid Low Band-Hi Pass)	160	20	2519	1	Hz
Frequency(4 Way-Mid Low Band-Low Pass	800	202	3174	1	Hz
Frequency(4 Way-Mid Hi Band-Hi Pass)	800	202	3174	1	Hz
Frequency(4 Way-Mid Hi Band-Low Pass)	4000	1008	20000	1	Hz
Frequency(4 Way-Hi Band)	4000	1008	20000	1	Hz

❖ Element Description

- Invert

  Invert the phase of the signal.

- Mode

  Invert the phase of the signal.

- Slope

  Slope of attenuation.

- Mute

  Slope of attenuation.

- Level

  Gain of signal.

- Frequency

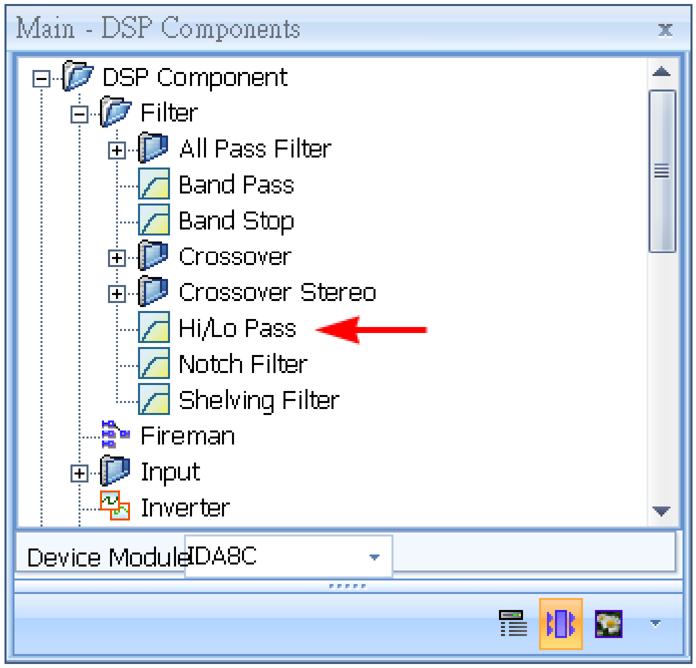
  Frequency of cut.

❖ Graphical Control Window

- Axis
  - X: Frequency of output signal(Hz)
  - Y: Level of input signal(dB)
- Control Points
  - Blue: Adjust Level and hi-pass Frequency or low-pass Frequency of mid band.
  - Yellow: Adjust Level and Both hi-pass Frequency and low-pass Frequency of mid band.
  - Red: Adjust Level and low band or hi band.

6.7.5 Hi/Lo Pass

❖Component Template

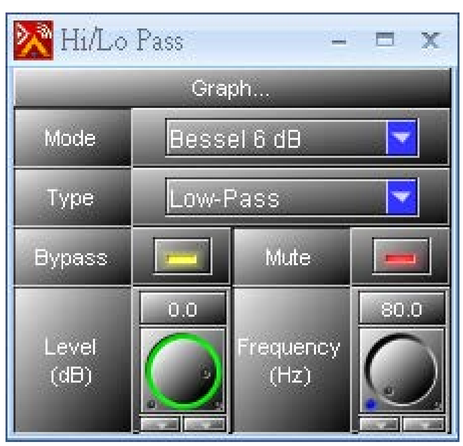


❖Component Appearance

❖Description

Hi/Lo Pass component passes high/lo frequencies and attenuates (i.e., reduces the amplitude of) frequencies lower/higher than its cutoff frequency.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mode	Bessel 6 dB	-	-	-	-
Type	Low-Pass	-	-	-	-
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
Level	0	-90.0	20.0	0.1	dB
Frequency	80	10	22000	1	Hz

❖ Element Description (todo)

- Mode

  Type of filtering.

- Type

  Low or High pass.

- Bypass

  Bypass the filter's module.

- Mute

  Bypass the filter's module.

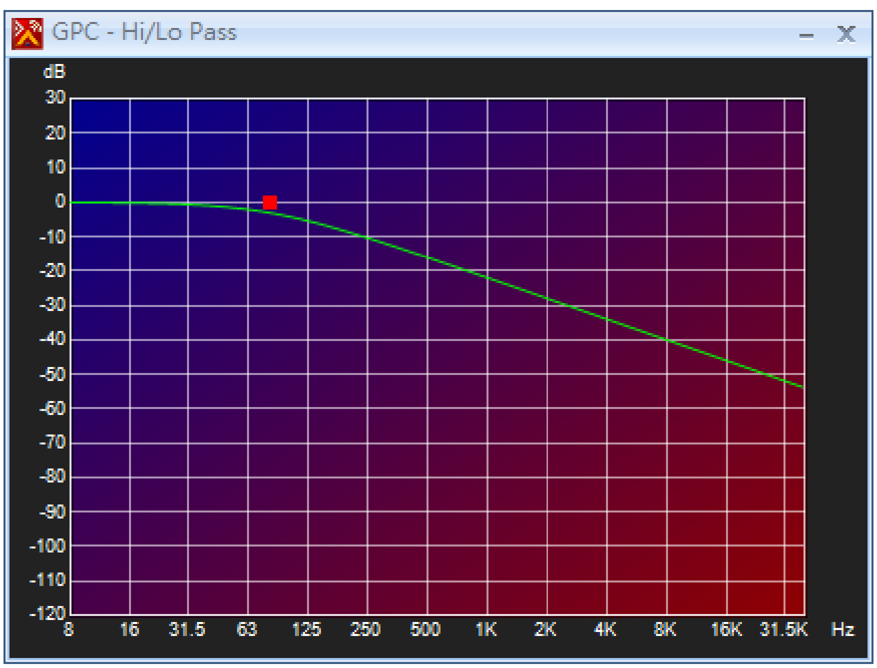
- Level

  Gain of signal.

- Frequency

  Frequency of cut.

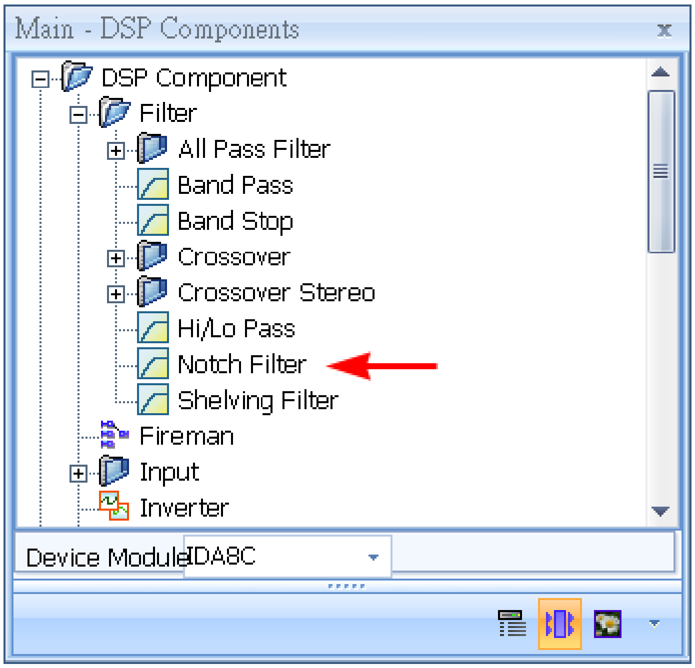
❖ Graphical Control Window



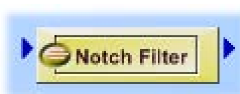
- Axis
  - X: Frequency of output signal(Hz)
  - Y: Level of input signal(dB)
- Control Points
  - Red: Adjust element Frequency(Horizontal) and Level(Vertical).

6.7.6 Notch Filter

❖Component Template



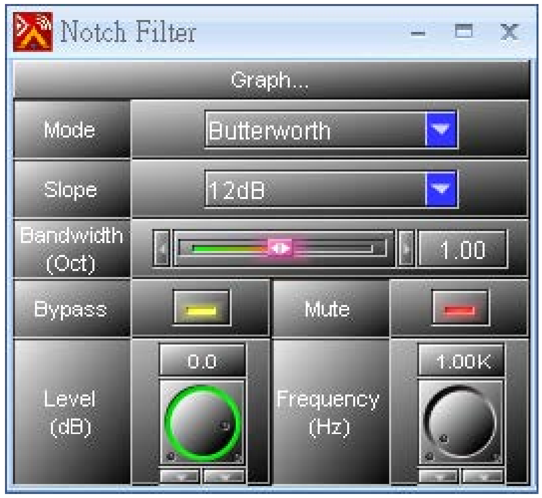
❖Component Appearance



❖Description

In signal processing, a band-stop filter or band-rejection filter is a filter that passes most frequencies unaltered, but attenuates those in a specific range to very low levels. It is the opposite of a band-pass filter. A notch filter is a band-stop filter with a narrow stop band (high Q factor).

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mode	Butterworth	-	-	-	-
Slope	12dB	-	-	-	-
Bandwidth	1	0.01	2.00	0.01	Octave
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
Level	0	-90.0	20.0	0.1	dB
Frequency	1000	10	22000	1	Hz

❖Element Description (todo)

- Mode

  The type of filer.

- Slope

  Slope of attenuation, the higher slope means the drop speeds level is faster.

- Bandwidth

  The difference between the upper and lower cut off frequencies in a contiguous set of frequencies.

- Bypass

  Bypass the filter's function.

- Mute

  Mute the audio signal.

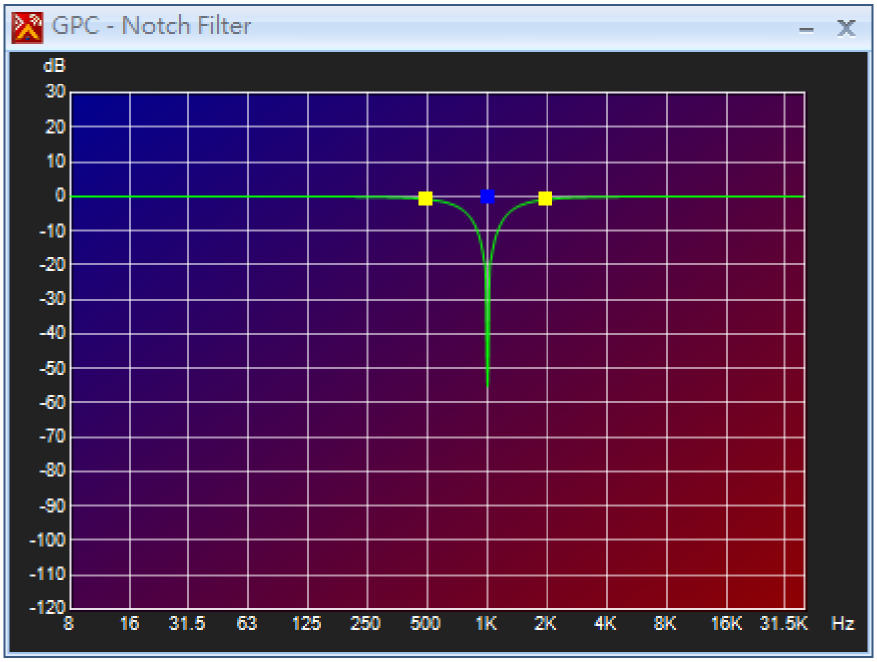
- Level

  Gain of audio signal.

- Frequency

The center frequency to be cut.

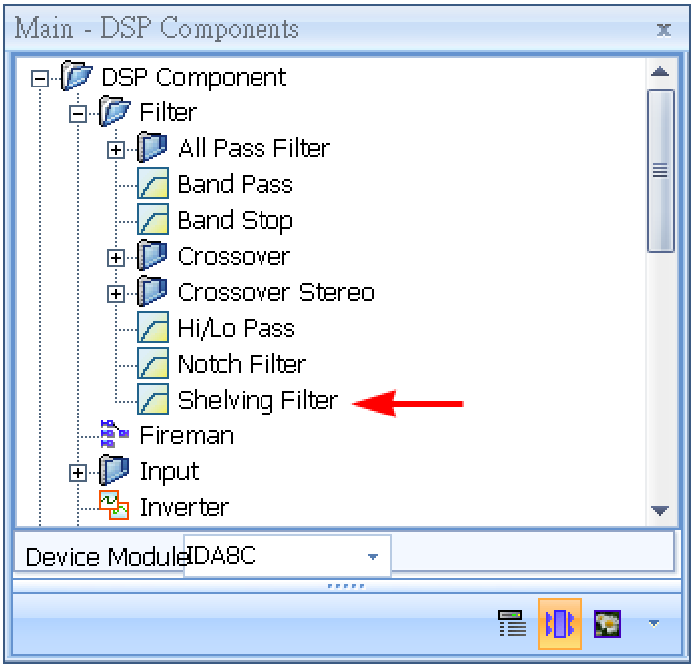
❖ Graphical Control Window



- Axis
  - X: Frequency of output signal(Hz).
  - Y: Level of input signal(dB).
- Control Points
  - Blue: Adjust element Frequency(Horizontal) and Level(Vertical).
  - Yellow: Adjust element Bandwidth.

6.7.7 Shelving Filter

❖Component Template

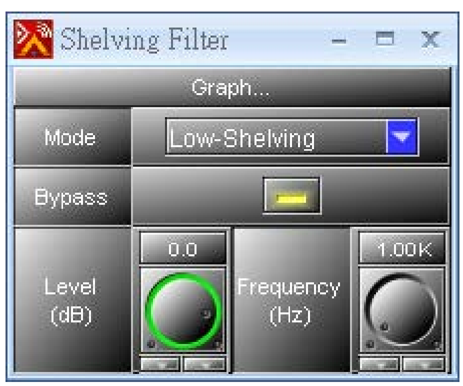


❖Component Appearance

❖Description

Shelving Filter component implements a first order response and provide an adjustable boost or cut to frequencies above or lower than a certain point.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mode	Low-Shelving	-	-	-	-

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Level	0	-90.0	20.0	0.1	dB
Frequency	1000	10	22000	1	Hz

❖Element Description (todo)

- Mode

  Type of filtering (high pass or low pass).

- Bypass

  Bypass the filter's function.

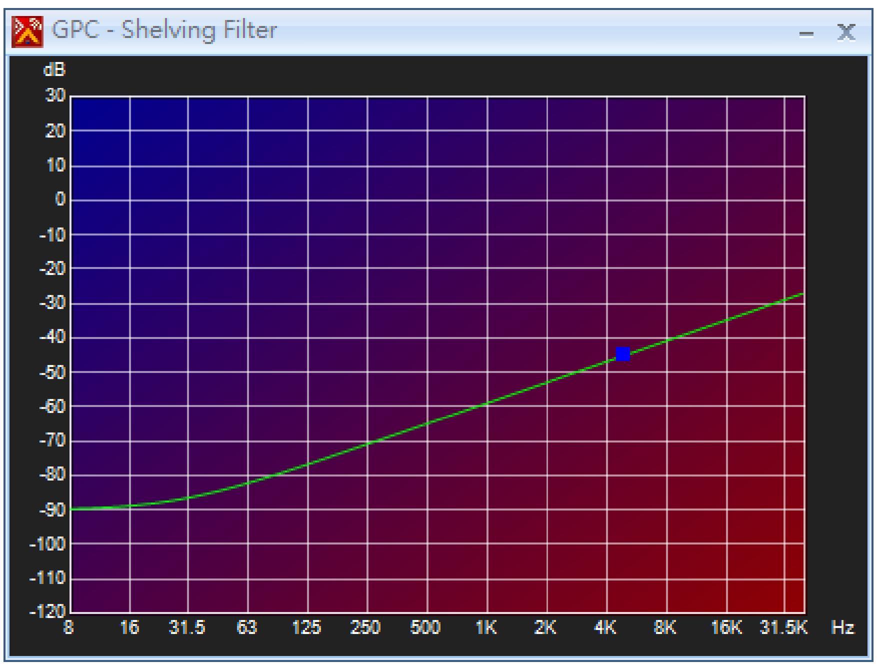
- Level

  The gain of audio signal.

- Frequency

  The frequency of cut.

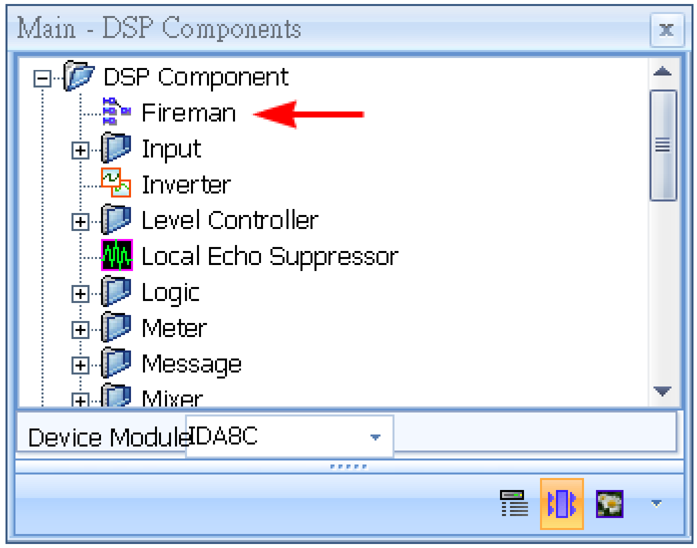
❖Graphical Control Window



- Axis
  - X: Frequency of output signal(Hz).
  - Y: Level of input signal(dB).
- Control Points
  - Blue: Adjust element Frequency(Horizontal) and Level(Vertical).

6.8 Fireman

❖ Component Template



❖ Component Appearance

❖ Description

This component represent Fireman Microphone with one channel audio output ready to further DSP processing.

❖ Control Window

❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB

❖ Element Description

- Mute

  Mute the audio signal.

- Level

Signal level of Fireman component.

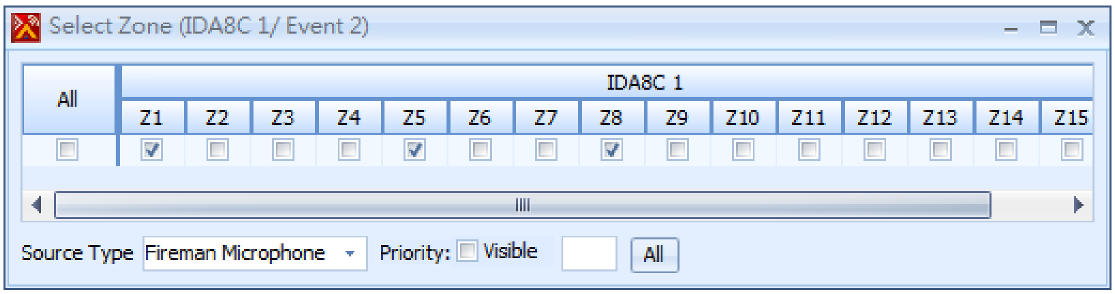
- Paging-Event Selection

Select paging event to bind with Fireman component. To do a paging using fireman microphone, it need to have a paging event bind with Fireman component.

Before assigning an paging event to the component, you need to create it first using settings window of Network Paging Component.

- Paging-Event Link Button

To open the settings window of binding paging event:

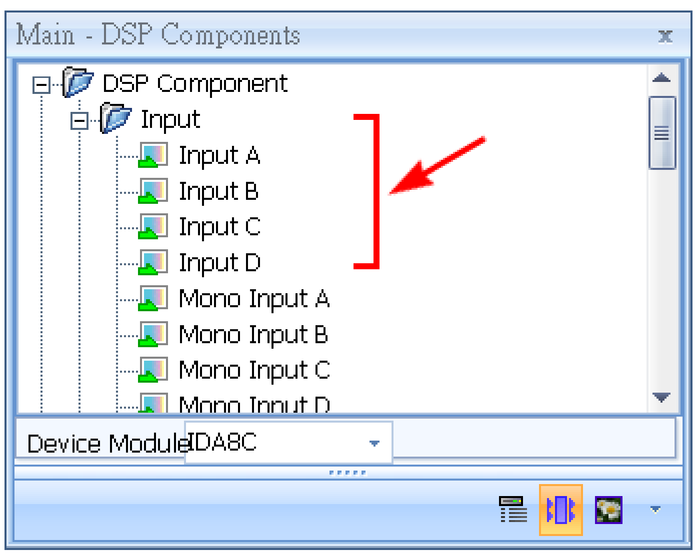


See more details in Network Paging topic.

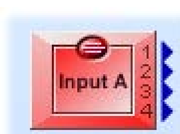
6.9 Input

6.9.1 Input

❖Component Template



❖Component Appearance



❖Description

Control the input signals parameters and monitor it. The analogue input modules represent the physical analogue inputs accessible on the rear panel of the Audio Processor. There are several separate modules for use with each of the audio board slots. Analogue input and output modules are added automatically to the DSP window of the relevant processor by defining their module configuration at the time you create them and should not be altered.

❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
Sensitivity	0dB	-	-	-	-
Phantom Power	Off	-	-	-	-
RTO	Off	-	-	-	-
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	todo	dB
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB

Name	Initial	Minimum	Maximum	Precision	Unit
Overload Thd.	0	-20.0	20.0	0.1	dB

❖Element Description

- Bypass

  Bypass audio of the channel.

- Mute

  Mute audio of the channel.

- Sensitivity

  Sets the input gain of the channel to preamplifier.

- Phantom Power

  Applies 48 VDC phantom power to the channel input for use with condenser mikes.

- RTO

  Indicates an input channel 'Routed To Output(s)'.

- Overload

  This LED light up if the signal of input channel greater than Overload Thd.

- Volume

  Meter showing the channel RMS level.

- Signal

  Indicates audio signal presence above -30 dB from chosen 'Sensitivity'.

- Level

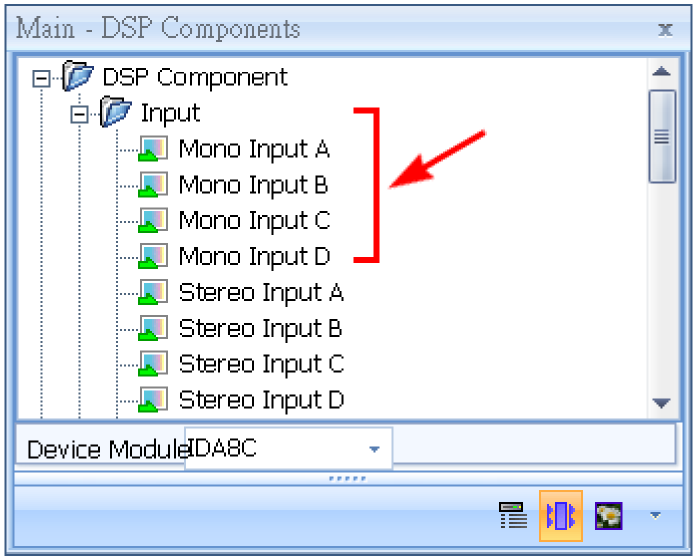
  Signal output level of the channel.

- Overload Thd.

  A threshold value to determine signal of a channel is overload or not.

6.9.2 Mono Input

❖ Component Template

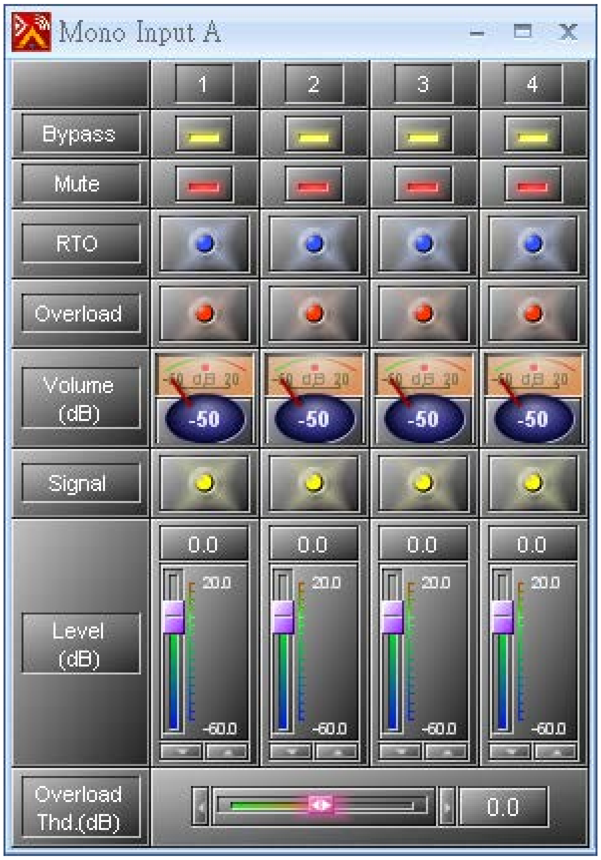


❖ Component Appearance

❖ Description

For AES/EBU input, the elements inside the component is very similar as Input component except without Sensitivity and Phantom Power.

❖ Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
RTO	Off	-	-	-	-
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	todo	dB
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload Thd.	0	-20.0	20.0	0.1	dB

❖Element Description

- Bypass

  Bypass audio of the channel.

- Mute

  Mute audio of the channel..

- RTO

  Indicates an input channel 'Routed To Output(s)'.

- Overload

  This LED light up if the signal of input channel greater than Overload Thd.

- Volume

  Meter showing the channel RMS level.

- Signal

  Indicates audio signal presence above -30 dB from chosen 'Sensitivity'.

- Level

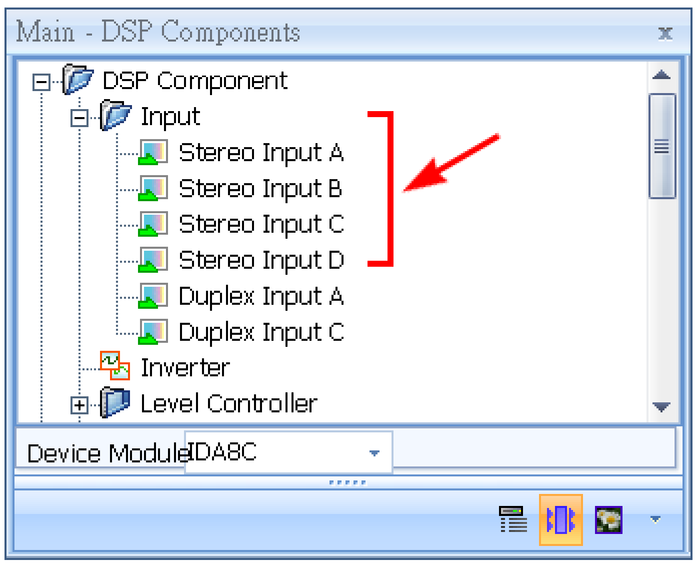
  Signal output level of the channel.

- Overload Thd.

  A threshold value to determine signal of a channel is overload or not.

6.9.3 Stereo Input

❖Component Template

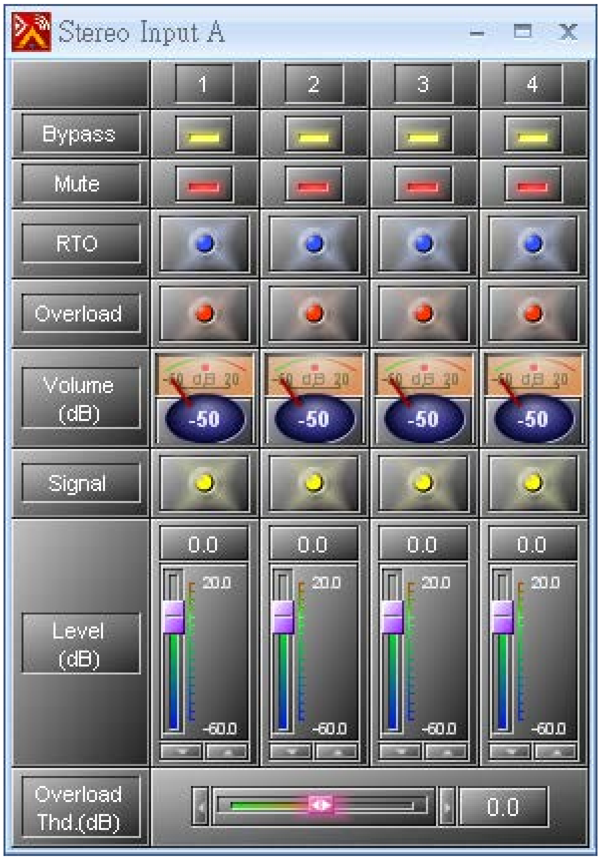


❖Component Appearance

❖Description

For AES/EBU input/output, the elements inside the component is very similar as Input component except without Sensitivity and Phantom Power. The audio input to the channel 1 of the board go to the component Stereo Input X channel 1 & 2. And board channel 3 to Stereo Input X channel 3 & 4.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
RTO	Off	-	-	-	-
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	todo	dB
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload Thd.	0	-20.0	20.0	0.1	dB

❖Element Description

- Bypass

  Bypass audio of the channel.

- Mute

  Mute audio of the channel..

- RTO

  Indicates an input channel 'Routed To Output(s)'.

- Overload

  This LED light up if the signal of input channel greater than Overload Thd.

- Volume

  Meter showing the channel RMS level.

- Signal

  Indicates audio signal presence above -30 dB from chosen 'Sensitivity'.

- Level

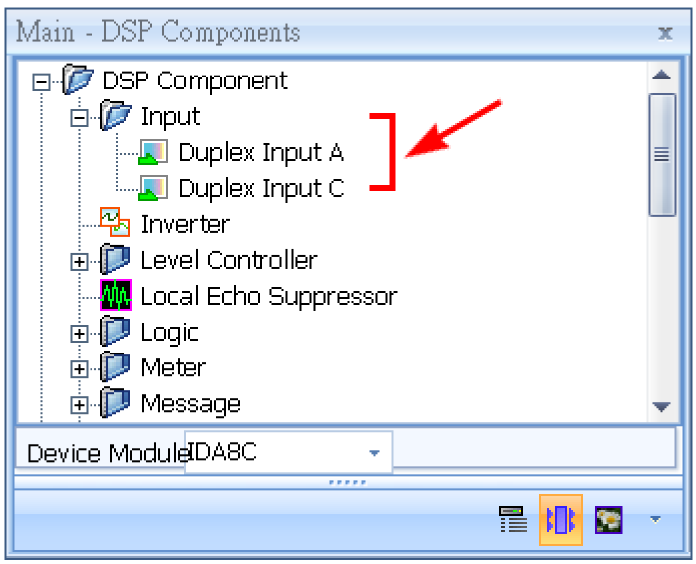
  Signal output level of the channel.

- Overload Thd.

  A threshold value to determine signal of a channel is overload or not.

6.9.4 Duplex Input

❖Component Template

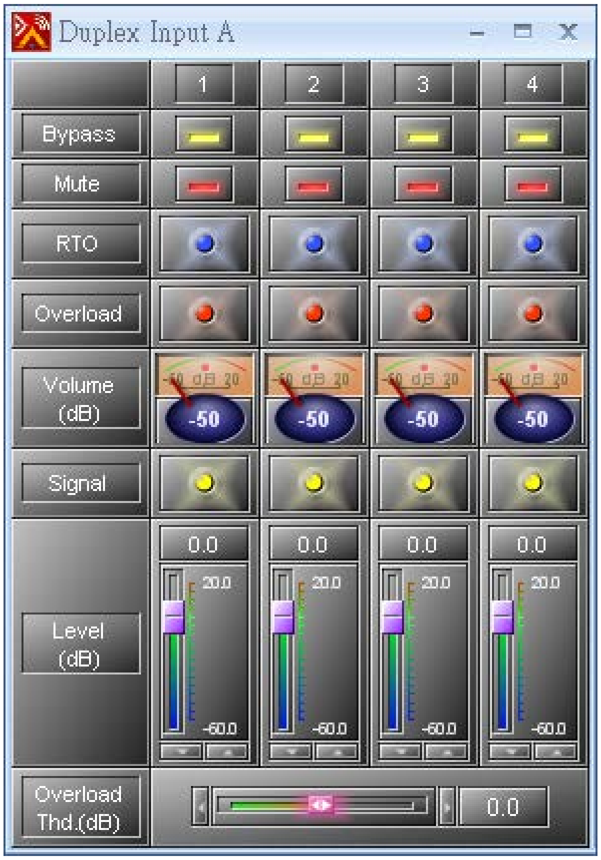


❖Component Appearance

❖Description

For AES/EBU input/output, the elements inside the component is very similar as Input component except without Sensitivity and Phantom Power. The audio input to the channel 1 of the board go to the component Duplex Input A channel 1 & 2, board channel 2 to Duplex Input A channel 3 & 4, board channel 3 to Duplex Output B channel 1 & 2,board channel 4 to Duplex Output B channel 3 & 4.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
RTO	Off	-	-	-	-
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	todo	dB
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload Thd.	0	-20.0	20.0	0.1	dB

❖Element Description

- Bypass

  Bypass audio of the channel.

- Mute

  Mute audio of the channel..

- RTO

  Indicates an input channel 'Routed To Output(s)'.

- Overload

  This LED light up if the signal of input channel greater than Overload Thd.

- Volume

  Meter showing the channel RMS level.

- Signal

  Indicates audio signal presence above -30 dB from chosen 'Sensitivity'.

- Level

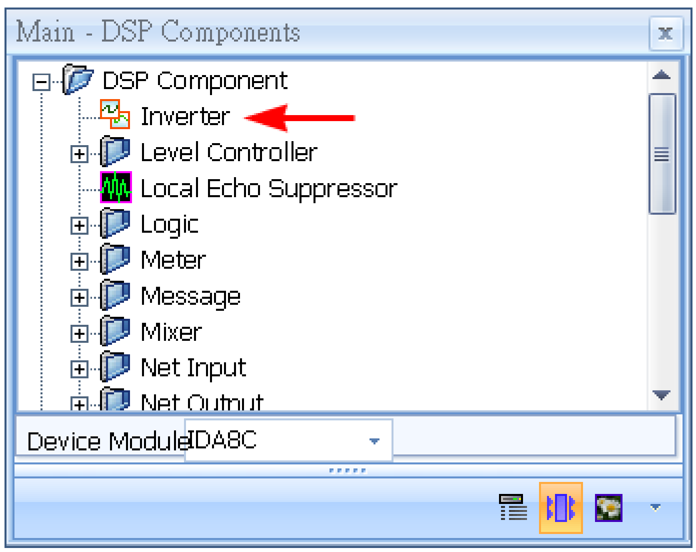
  Signal output level of the channel.

- Overload Thd.

  A threshold value to determine signal of a channel is overload or not.

6.9.5 Inverter

❖Component Template



❖Component Appearance

❖Description

This component change the signal phase of 90 degrees.

❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	Off	-	-	-	-
Invert	Off	-	-	-	-

❖Element Description

- Mute

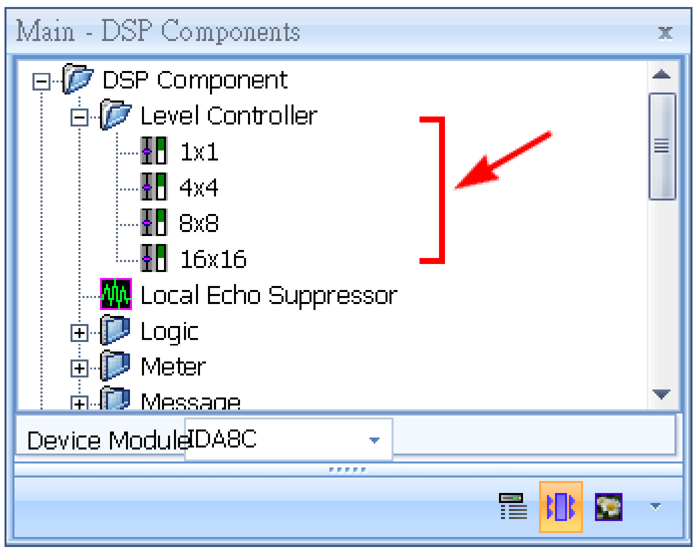
  Mute the inverter module.

- Invert

  Active the inverter.

6.10 Level Controller

❖Component Template

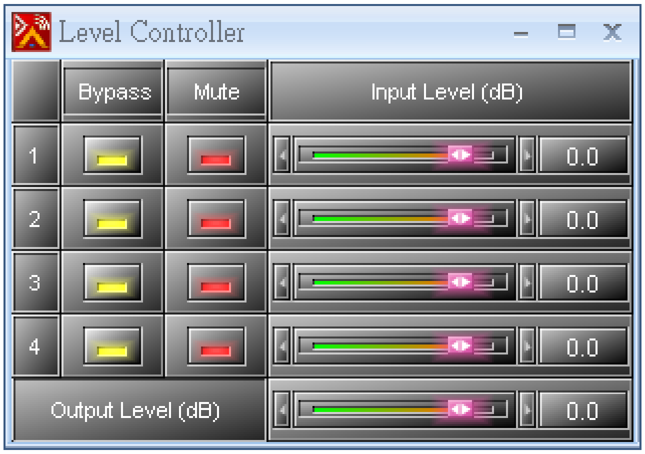


❖Component Appearance

❖Description

This component can adjust the signal level.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
Input Level	0	-90.0	20.0	0.1	dB
Output Level	0	-90.0	20.0	0.1	dB

❖Element Description

- Bypass

  Bypass the module's function.

- Mute

  Mute the sound coming out from the component.

- Input Level

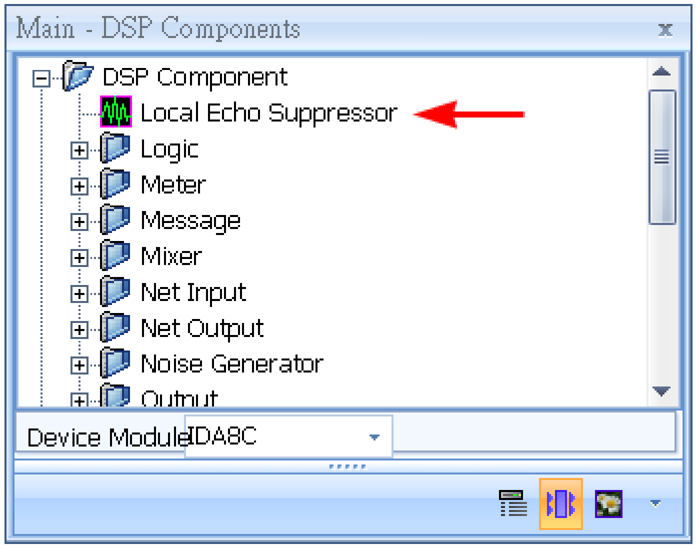
  Adjust the level of an input channel.

- Output Level

  Adjust the level of output.

6.11 Local Echo Suppressor

❖Component Template

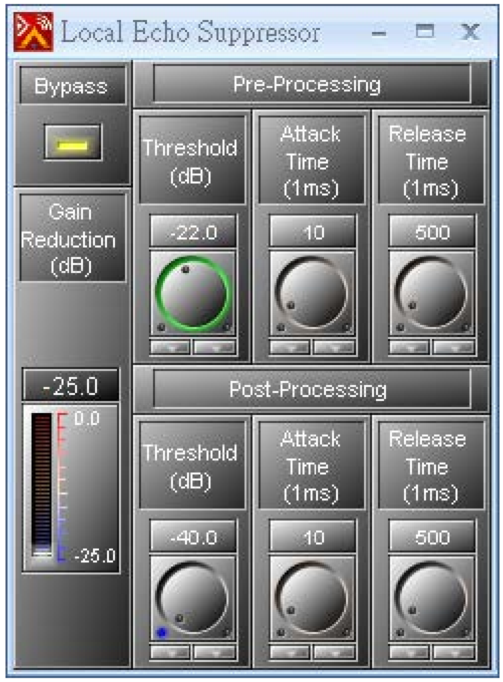


❖Component Appearance

❖Description

This component is designed for suppressing local Echo or feedback. It's shared in two parts, pre-processing and post-processing. The Pre-Processing reduces the level above the threshold. This is to avoid feedback. The Post-Processing removes the echo below a threshold. This is to avoid the residual echo.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Gain Reduction	0	-25.0	0	0.1	dB
Pre-Processing Threshold	-22.0	-40.0	0	0.1	dB
Pre-Processing Attack Time	10	1	100	1	ms
Pre-Processing Release Time	500	100	5000	1	ms
Post-Processing Threshold	-40.0	-40.0	-20.0	0.1	dB
Post-Processing Attack Time	10	1	100	1	ms
Post-Processing Release Time	500	100	5000	1	ms

❖Element Description

- Bypass

  Switches off the component operations.

- Gain Reduction

  Display the gain of output volume.

- Pre-Processing Threshold

  When input volume is bigger than Pre-Processing Threshold ,it will Reduce the volume to Threshold.

- Pre-Processing Attack Time

  The time needed to change the volume from original level to the expected level.

- Pre-Processing Release Time

  The time needed to change the volume from original level to the expected level.

- Post-Processing Threshold

  When input volume is smaller than Post-Processing Threshold ,it will suppress the residue echo sound.

- Post-Processing Attack Time

  The time needed to change the volume from original level to the expected level.

- Post-Processing Release Time

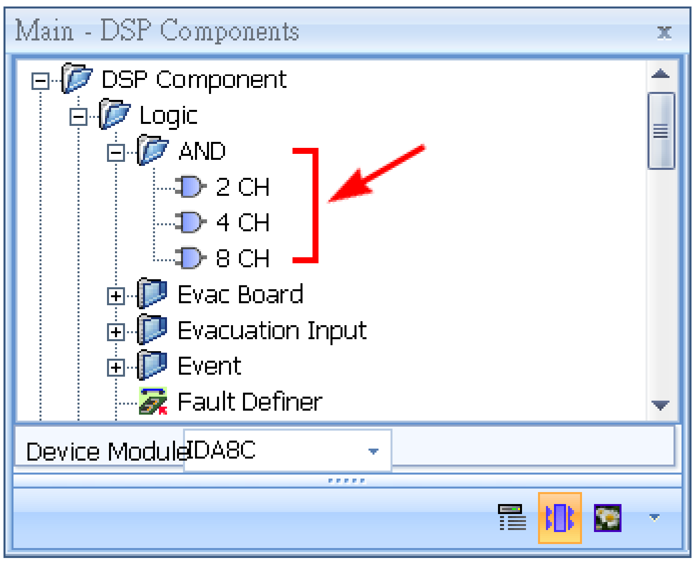
  The time needed to change the volume from expected level to the original level.

6.12 Logic

Components process logic signal listed inside this section.

6.12.1 AND

❖Component Template



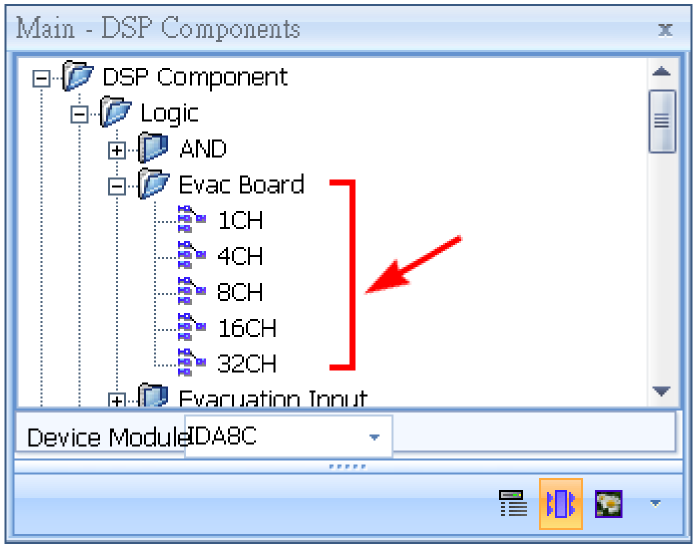
❖Component Appearance

❖Description

This component allows the logical operation "AND", applied on logic signal.

6.12.2 EVAC Board

❖ Component Template



❖ Component Appearance

❖ Description

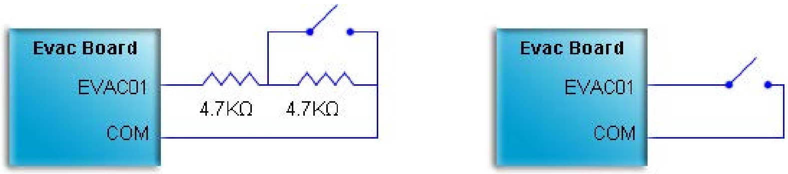
This component is corresponds to URGP peripheral devices, it extend audio processor's logic input channel.

❖ Control Window

❖ Parameters

- Monitor

  Enable or disable external device monitoring. If this option is enabled, it need two 4.7k Ohm resisters in external device circuit.



Monitoring Mode Non-Monitoring Mode

- Type

  To specify mode of activate, there are two chooses:

  - N.O.

    Normally open, close the contact to output signal.

  - N.C.

    Normally close, open the contact to output signal.

- Status

  Shows states of external device. There are four states:

  - Open

    A open circuit of external device is detected.

  - Short

    A short circuit of external device is detected.

  - Activate

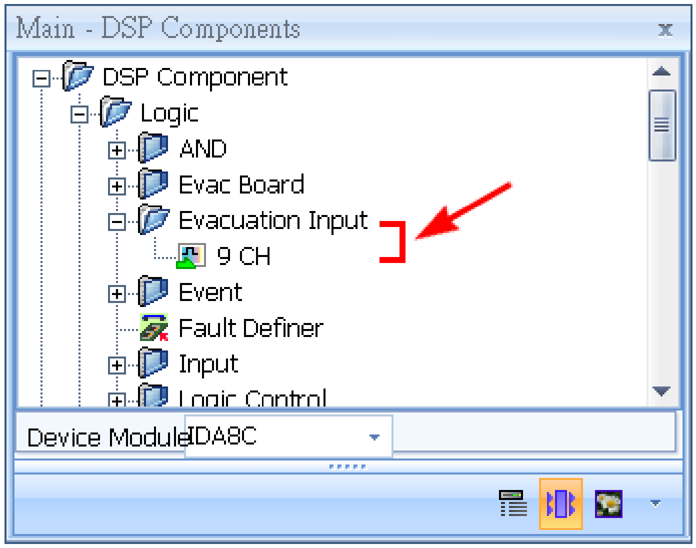
    Logic output is activate.

  - Deactivate

    Logic output is deactivate.

6.12.3 Evacution Input

❖ Component Template

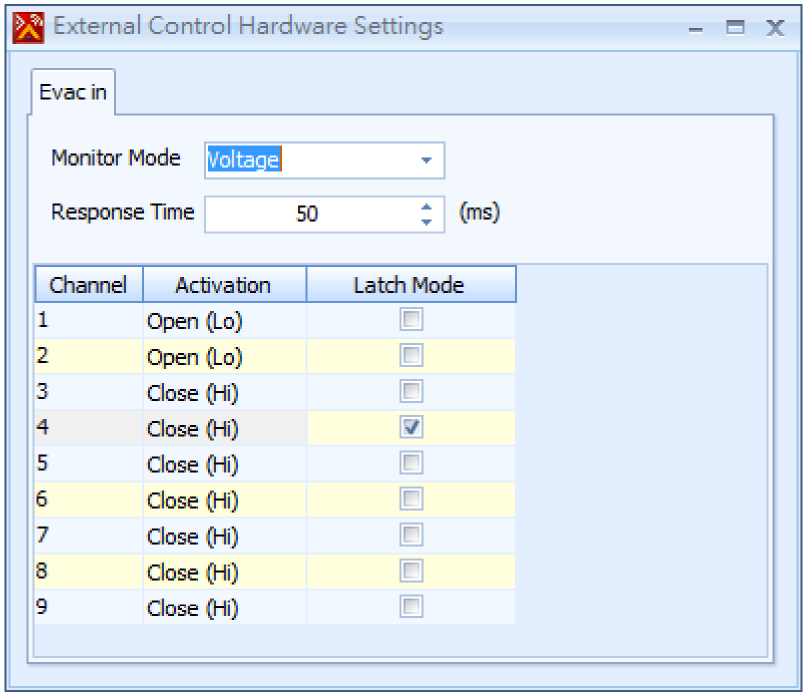


❖ Component Appearance

❖ Description

This component corresponds to the Evacuation Input of IDA8C/S, output logic signal be the other logic component's input.

❖ Control Window



Parameters

- Monitor Mode

  There are two options:

  ○ Voltage:

  Logic output of component trigger by a voltage polarization change.

  ○ Contact

  Logic output of component trigger by a open or close the pin +/- of connector in rear panel.

- Response Time

  Determine how quick of the system response for changing of hardware status from Open(Lo) to Close(Hi) or from Close(Hi) to Open(Lo).

- Activation

  Specify this parameter to determine what hardware condition generate 1 output. There are two chooses:

  ○ Open(Lo): Component output 1 when contact open(contact mode) or voltage low(voltage mode), output 0 vise versa.

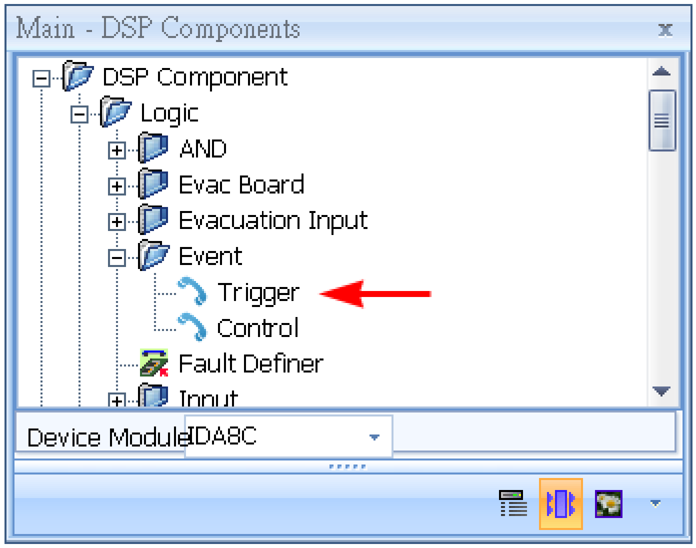
  ○ Close(Hi): Component output 1 when contact close(contact mode) or voltage hi(voltage mode), output 1 vise versa.

6.12.4 Event

The components work with events will be listed in this section.

6.12.4.1 Trigger

❖ Component Template



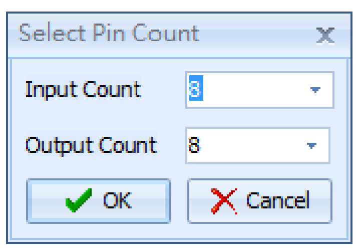
❖ Component Appearance

❖ Description

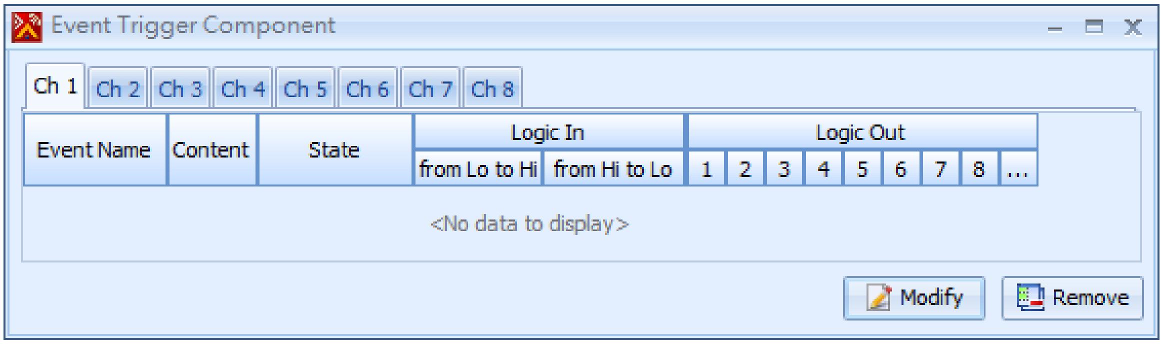
This component link events to logic signal, linked events will be triggered if the input logic signal of component is from lo to hi or hi to lo depending on settings.

❖ Control Window

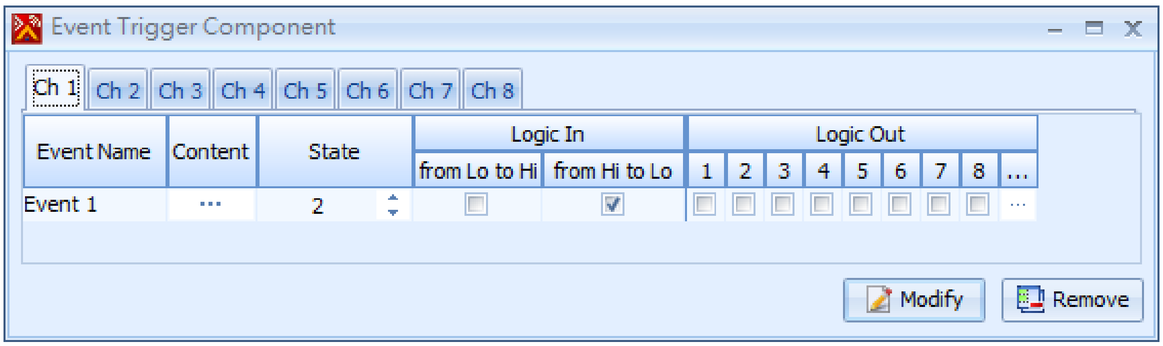
When component is drag from the component template and drop to the graphic editor for DSP design, a window show to specify the input and output channel counts:



Click the component the control window shows, there are several tabs on the top of window, each tab is for a logic input channel:



After events is linked to the component the control window list it in grids:



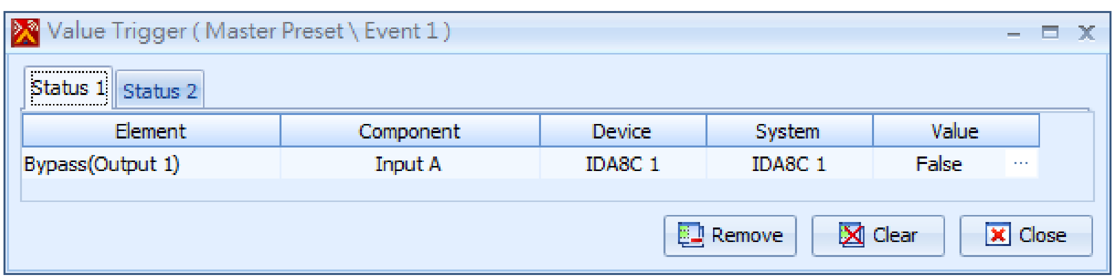
❖ Parameters

- Event Name

  Show the name of event linked.

- Content

  Open the event settings window.



- State

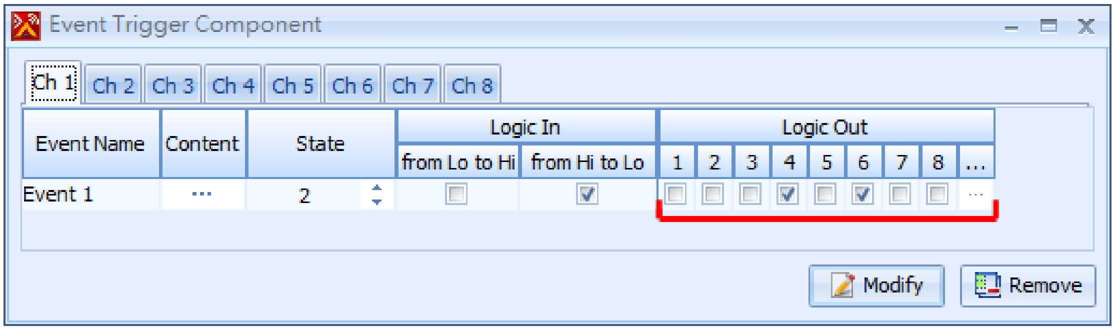
  Specify the state number for triggering an event.

- Logic In

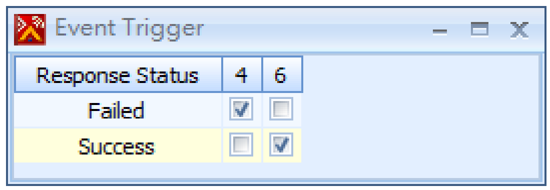
  Define which condition triggers linked events.

- Logic Out

  Define output signal state after linked event executed, basically there are two cases need to considerate: event execute success and failed. To setup this configuration you need select the working output signal first:



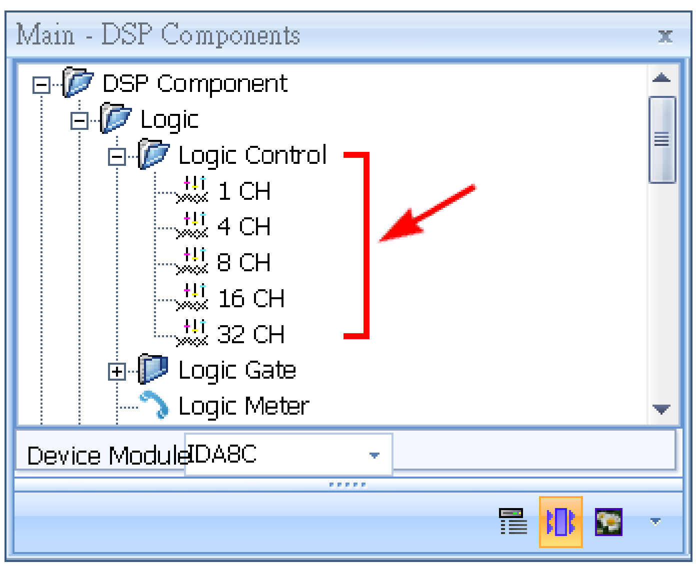
In above figure, the logic output channel 4 and 6 are selected. to detail settings click the grid cell [...]:



In above figure, the settings means when Event 1 is Failed, output 4 will be 1 and output 6 will be 0, and when Event1 is successfully executed the output 4 will be 0, output 6 will be 1.

6.12.5 Logic Control

❖Component Template

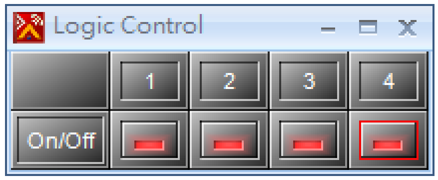


❖Component Appearance

❖Description

This component allow user to generate logic signal using button in software. For each channel there is an element for control logic signal for output.

❖Control Window



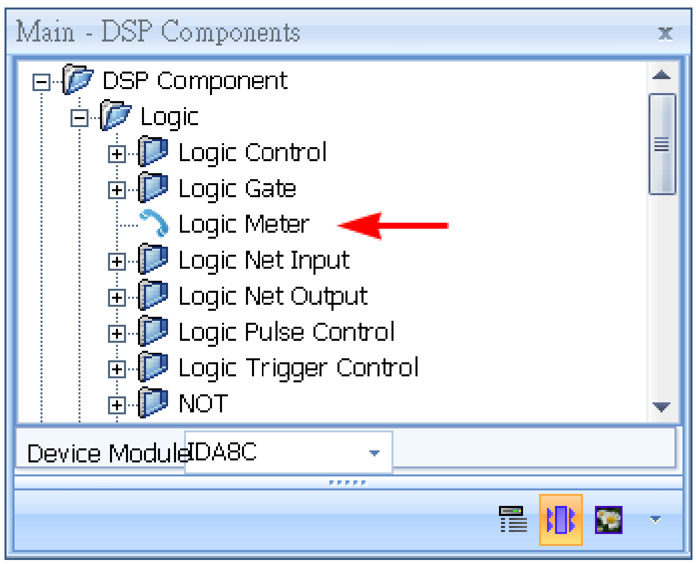
❖Elements

- On/Off

  To output logic signal 0 or 1.

6.12.6 Logic Meter

❖Component Template

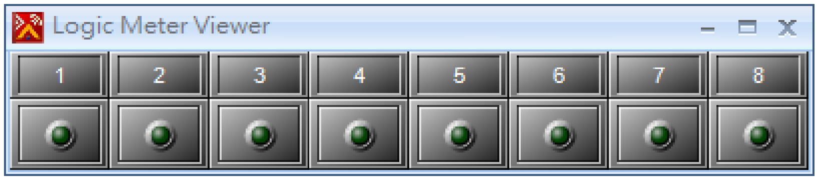


❖Component Appearance

❖Description

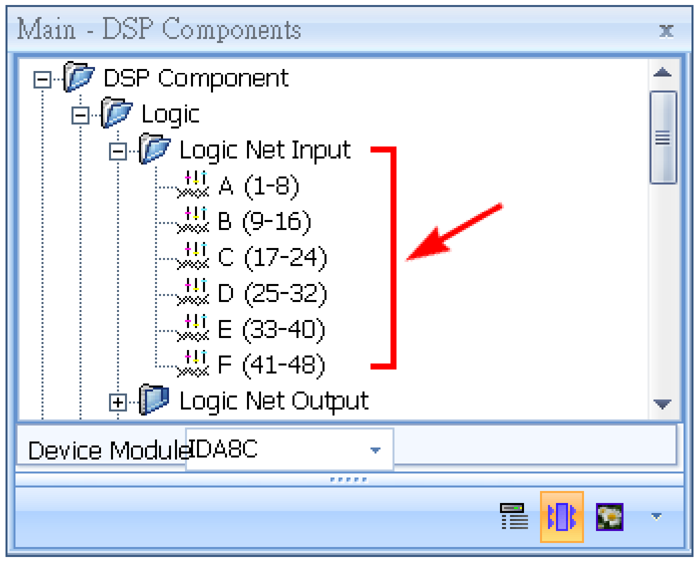
This component have LEDs to indicate state of logic signal.

❖Control Window



6.12.7 Logic Net Input

❖Component Template



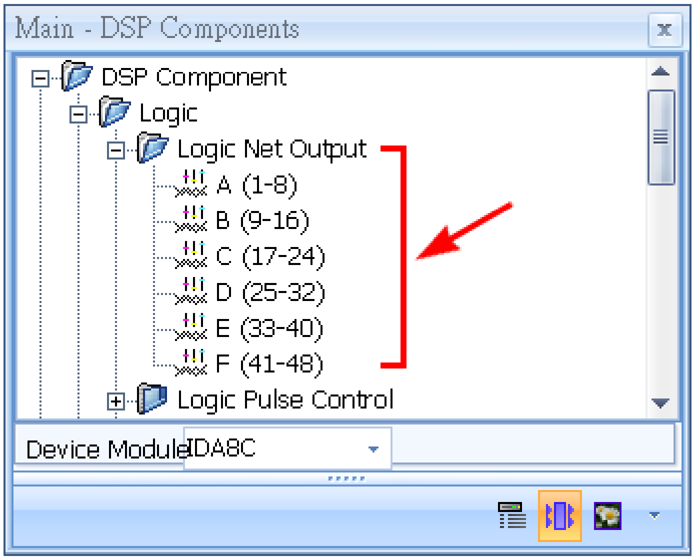
❖Component Appearance

❖Description

This component transfer logic signals over Ateis Net.The logic signal goes into the Logic Net Out component in one device using Ateis Net card, and come out to Logic Net In component in another device using Ateis Net card as well. The signal path is one to many relationship, There are 48 channels available to carry signals, they are numbered and identified by the number. For example, a signal goes into channel 1 of component Logic Net Out A, you can get it from the component Logic Net In A channel 1.

6.12.8 Logic Net Output

❖Component Template



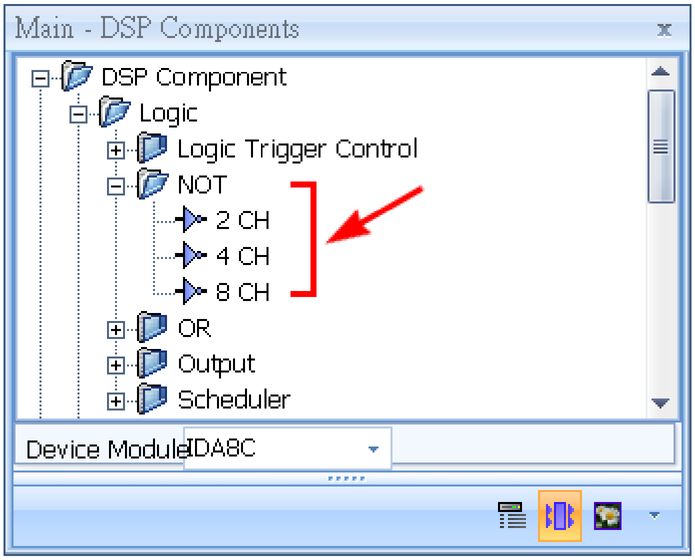
❖Component Appearance

❖Description

This component transfer logic signals over Ateis Net.The logic signal goes into the Logic Net Out component in one device using Ateis Net card, and come out to Logic Net In component in another device using Ateis Net card as well. The signal path is one to many relationship, There are 48 channels available to carry signals, they are numbered and identified by the number. For example, a signal goes into channel 1 of component Logic Net Out A, you can get it from the component Logic Net In A channel 1.

6.12.9 NOT

❖Component Template



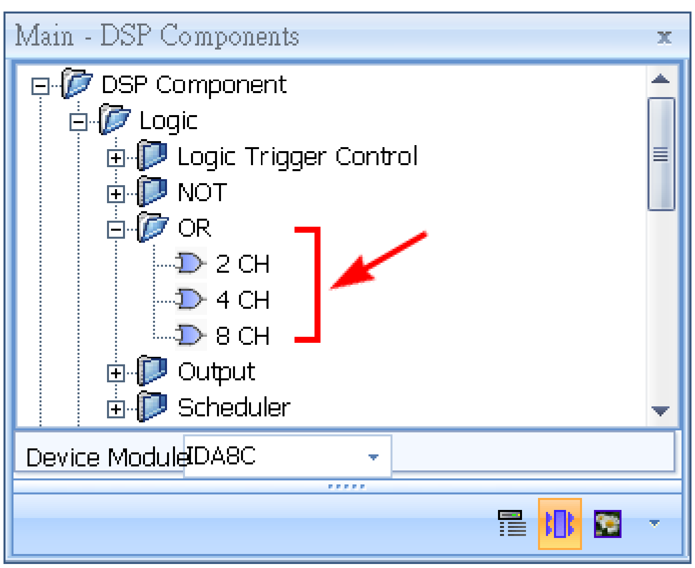
❖Component Appearance

❖Description

This component performs logic not operation.

6.12.10 OR

❖Component Template



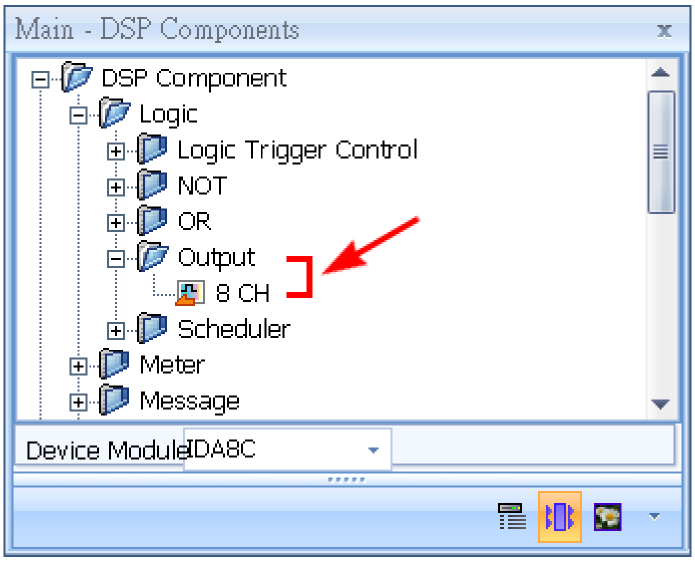
❖Component Appearance

❖Description

This component performs logic or operation.

6.12.11 Output

❖Component Template



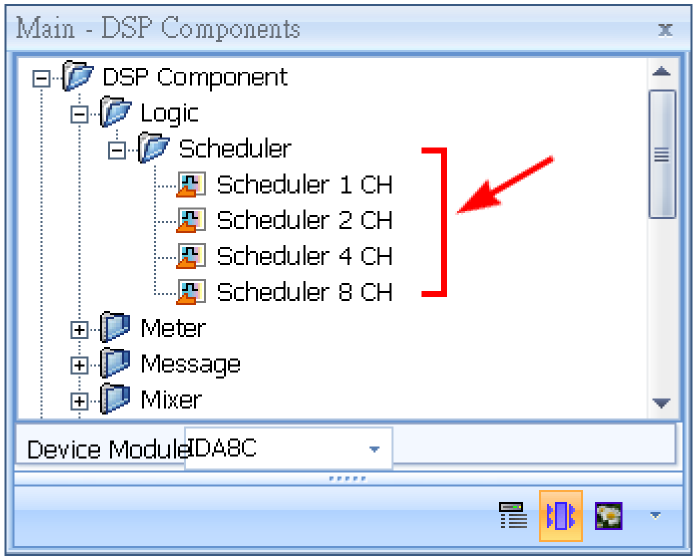
❖Component Appearance

❖Description

This component is corresponding to contact outputs.

6.12.12 Scheduler

❖Component Template



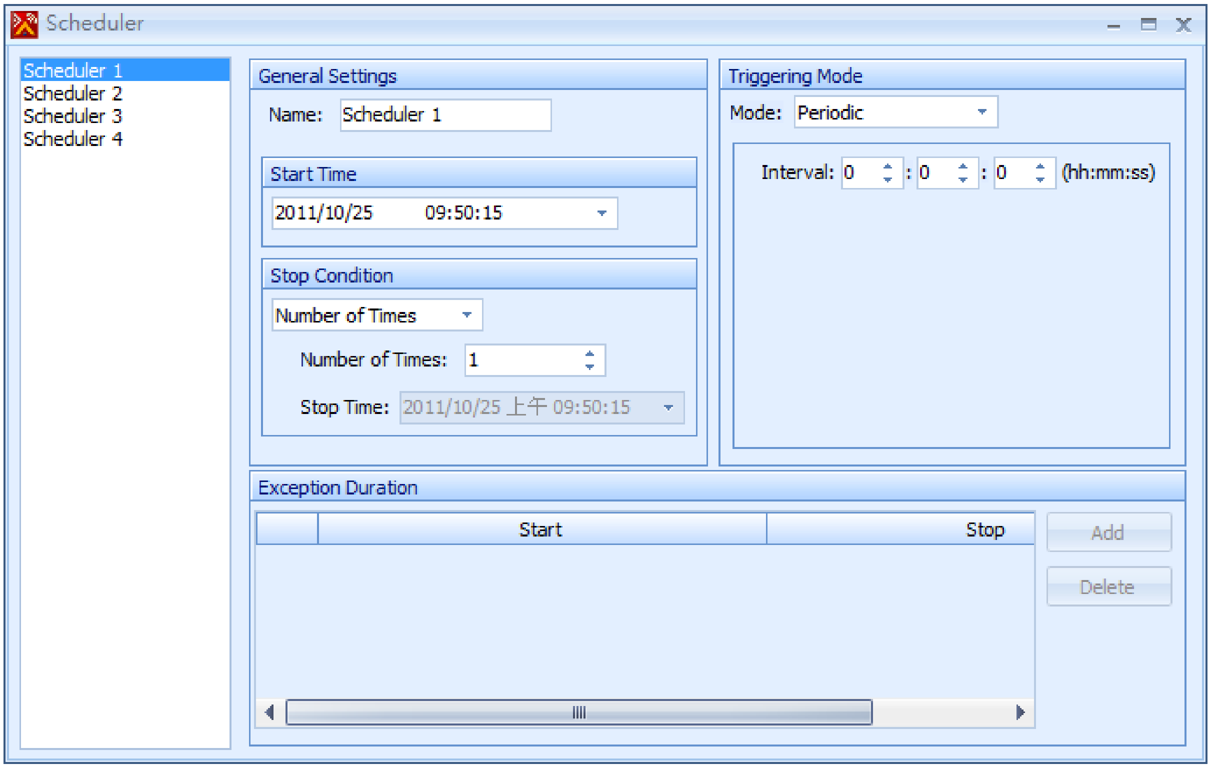
❖Component Appearance

❖Description

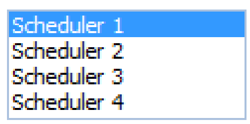
This component allow user to arrange the timing of generating logical pulse to output. The length of pulse is 1 sec. There are four types of scheduler: Scheduler 1 CH, Scheduler 2 CH, Scheduler 4 CH, Scheduler 8 CH. For each channel, there is s scheduler working on it.

The pins on top side enable(signal = 1) or disable(signal = 0) the scheduler belong to the channel pin with. the down pins generate pulse signal depends on the settings of component.

❖Control Window

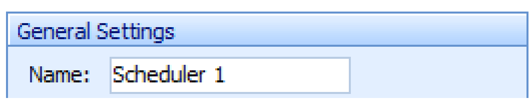


❖Scheduler List



On the left side of window, there is a box to list all channels of scheduler. Click it can change the focus of scheduler and the right part of window which is the settings of the scheduler will switch to corresponding scheduler.

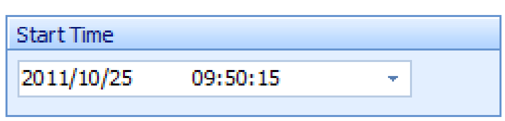
- General Settings



  - Name

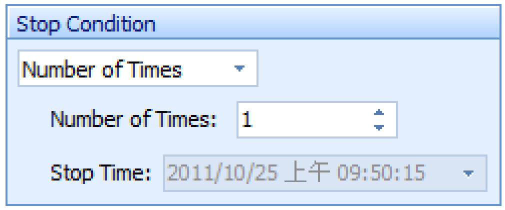
    The name of scheduler.

❖Start Time



The start time of the scheduler,

❖Stop Condition



Specify the condition to deactivate scheduler, there are three kinds of stop condition:

- Continue

  In this option, scheduler never stop, triggering mode settings is valid from now and forever.

- Number of Times

  Scheduler triggers N times using triggering mode settings, where N can be set on Number of Times parameter.

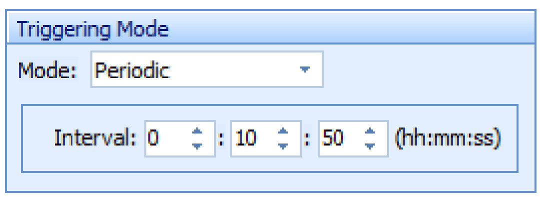
- Stop Time

  Scheduler is active from start time to the time specified in Stop Time parameter.

❖ Triggering Mode

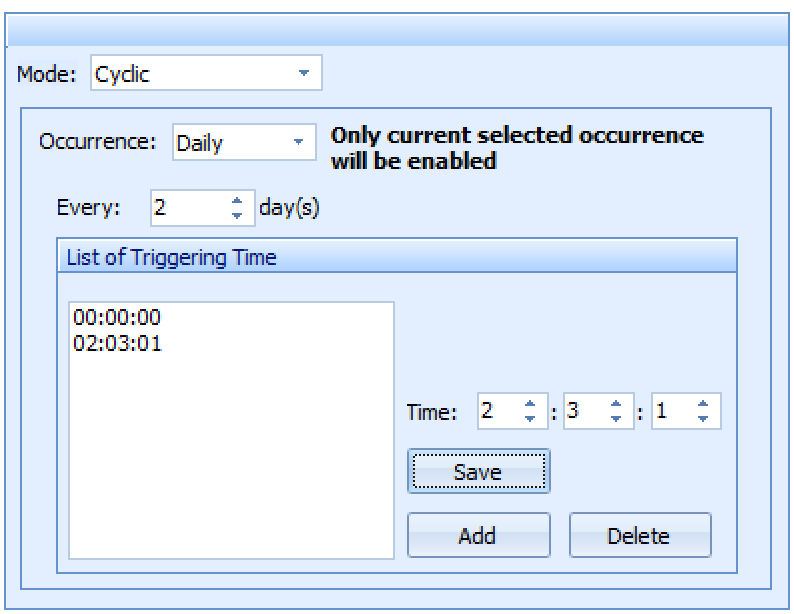
The triggering mode define the rule of trigger an pulse to output.

- Periodic



  The interval is the time between two triggers.

- Cyclic



The Occurrence parameter determines the cycle type of triggering, there are five types available: Hourly/Daily/Weekly/Monthly/Yearly. It allow user to define the interval between cycles by specify the parameter of Every N cycle units, where cycle unit could be Hour/Day/Week/Month/Year.

In the box of List of Triggering Time, you can add or remove the triggering time in the cycle.

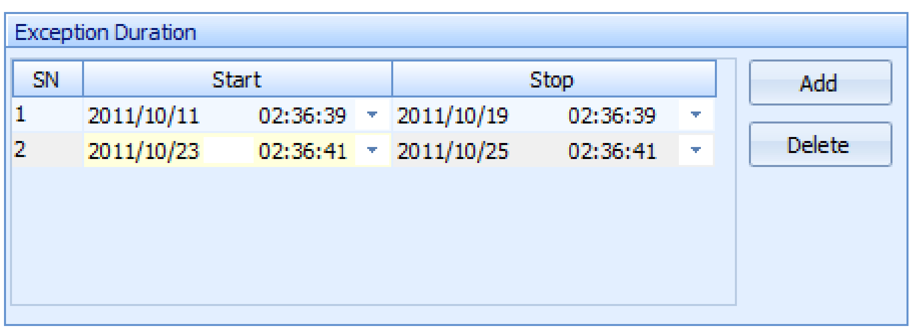
- By Excel File

	A	B	C	D	E	F	G	H
1	2011.1	24HR						
2	1	08:00:00	09:00:00	10:00:00	11:00:00	12:00:00	13:00:00	14:00:00
3	2	08:00:00	09:00:00	10:00:00	11:00:00	12:00:00	13:00:00	14:00:00
4	3	08:00:00	09:00:00	10:00:00	11:00:00	12:00:00	13:00:00	14:00:00
5	4	08:00:00	09:00:00	10:00:00	11:00:00	12:00:00	13:00:00	14:00:00
6	5	08:00:00	09:00:00	10:00:00	11:00:00	12:00:00	13:00:00	14:00:00
7	6	08:00:00	09:00:00	10:00:00	11:00:00	12:00:00		

It allow user specify triggering times using an excel file, the rules are listed below:

- Field A1 is the Year.Month to scheduling.
- B1 ~ X1 is the field for time format, there are two choose:
  - 24HR
  - AM/PM
- A2 ~ A 32 represent the day of a month.
- For each row, B to X fields are the time to be trigged.

❖ Exception Duration

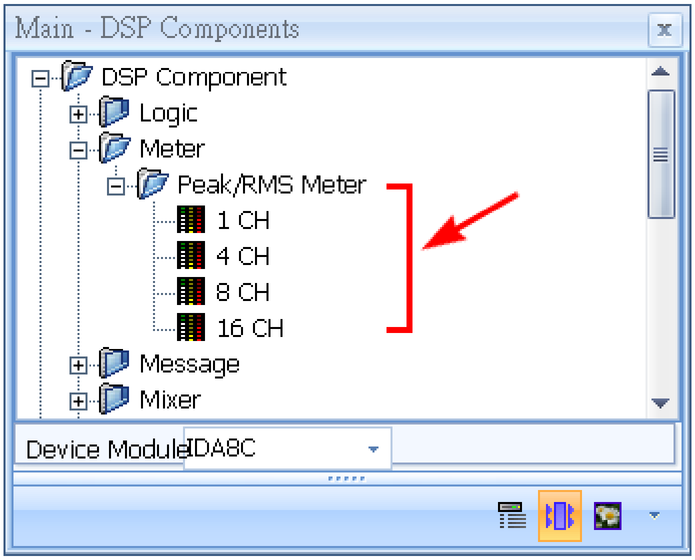


Event the mechanism for triggering time setup is very flexible, but there still always need the exceptions. On the lower part of scheduler settings window, A block Exception Duration allow user to edit a list for the exceptions of the triggering, each exception consist of a start time and stop time. During the exception time, all triggering will be masked.

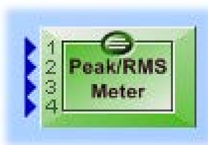
6.13 Meter

6.13.1 Peak/RMS Meter

❖Component Template



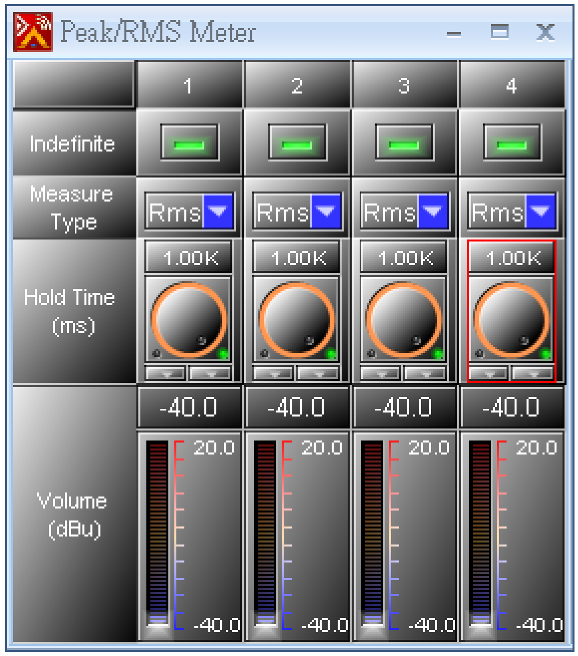
❖Component Appearance



❖Description

This component display the amplitude of the audio signal coming in. There are two approaches available for measuring the signal. The first one is RMS, which is offend use to measure complex waveforms, especially non-repeating signals like noise. The second is Peak approach.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Indefinite	Off	-	-	-	-
Measure Type	RMS	-	-	-	-
Hold Time	10	10	1000	10	ms
Volume	-40.0	-40	20	0.1	dBu

❖Element Description

- Indefinite

  Block the display of level value.

- Measure Type

  Type of measure.

- Hold Time

  Able to see keep displayed the value along a the hold time.

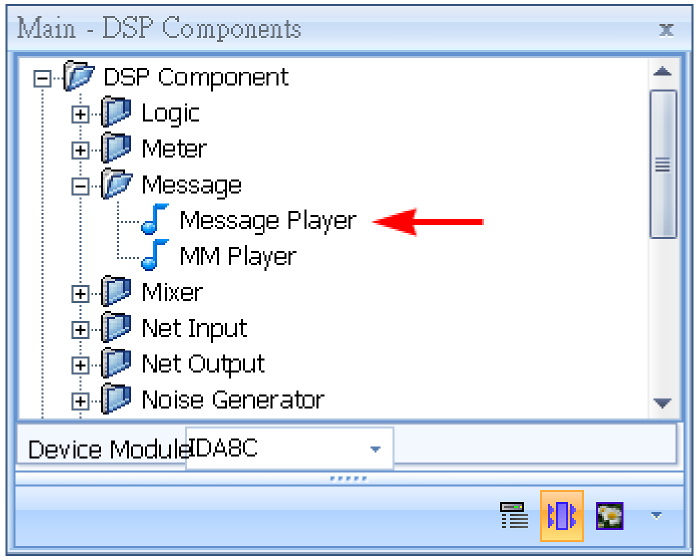
- Volume

  Measured value of signal.

6.14 Message

6.14.1 Message Player

❖ Component Template



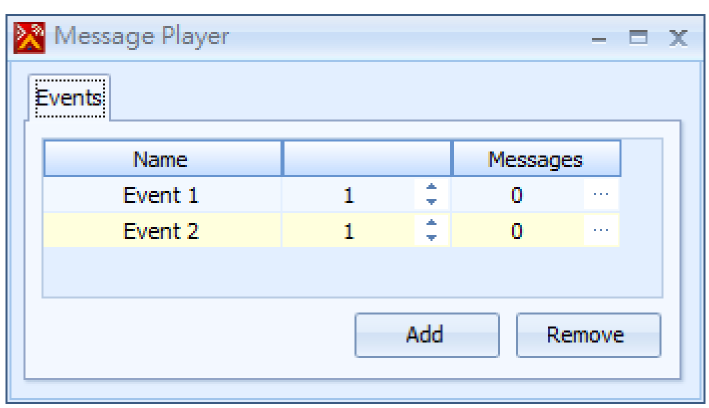
❖ Component Appearance

❖ Description

This component play the audio message files which stored in Ateis devices. It is able to have multiple events binding with a message in the event list. each event can be assigned to a source to trigger it. If more than one event are triggered in the same message player component. The event with the highest priority will play the message.

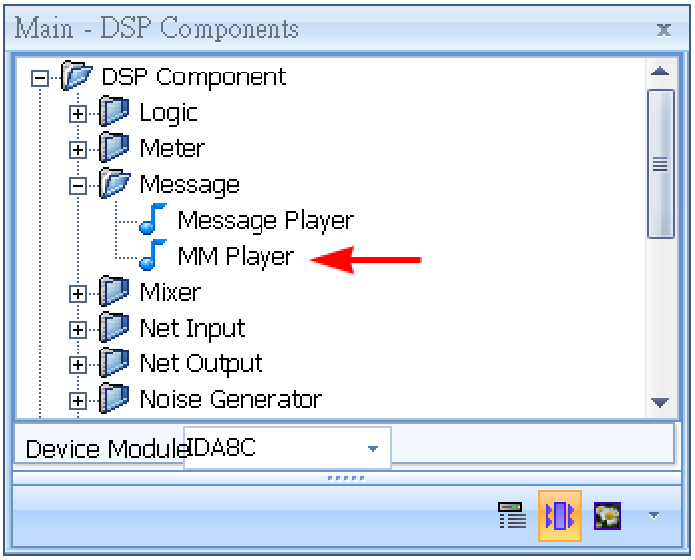
⚠ The smaller value with higher priority.

❖ Control Window



6.14.2 MM Player

❖Component Template



❖Component Appearance

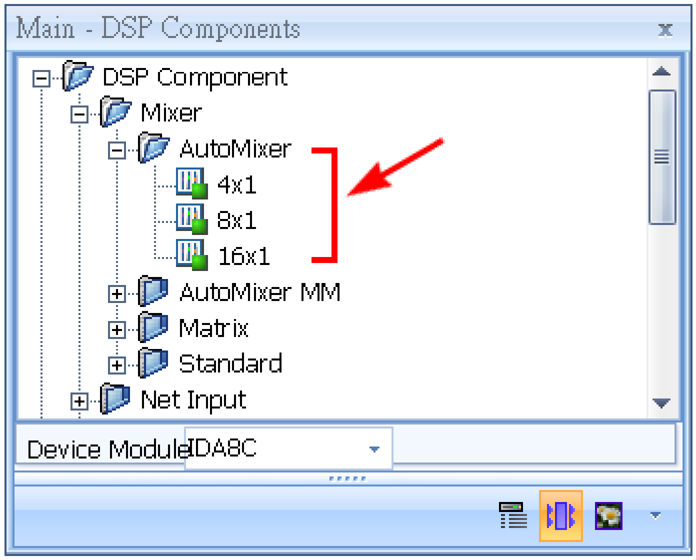
❖Description

This component play the messages stored in Ateis device. It allow user to specify playing sequence of message and is possible to play message by logic signal input of the component.

6.15 Mixer

6.15.1 AutoMixer

❖Component Template



❖Component Appearance

❖Description

This component Allows mix several signals to one output. The Activation of channel(s) can be done manually or automatically by signal and priority.

The auto mixer adjusts the signal output level on the output node depending on the number of activated input channels. This function can be useful in conference applications or houses of workshop, where usually unattended mixing has to be executed.

A special feature of this component is the priority function, which can be used for auto-paging applications or to implement a voice triggered chairman's mike in a small conference system.

❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Type	Manual	-	-	-	-
Last Mic. On Designated	Channel 1	-	-	-	-
Mute	On	-	-	-	-
Invert	Off	-	-	-	-
Solo	Off	-	-	-	-
Gate On	Off	-	-	-	-
Input Level	0	-90.0	20.0	0.1	dB
Manual mode	On	-	-	-	-
Automatic mode priority level	1	2	3	4	-
Output Mute	Off	-	-	-	-
Output Level	0	-90.0	20.0	0.1	dB
Threshold	-40.0	-90.0	20.0	0.1	dB
Background Threshold	15.0	0	100.0	0.1	dB
Hold Time	5	0	10.0	0.1	Second
Attenuation Response Time	10	10	10000	1	ms
NOM Attenuation Step	0	0	6	0.1	dB
NOM Attenuation Max	0	0	100	0.1	dB
Attenuation Gain	-100.0	20.0	-100.0	0.1	dB
Open Mic. Limits	No limit	-	-	-	-
NOM Attenuation Mode	1: (NOM-1) *step	-	-	-	-

❖Element Description

- Type

  Select the mode, manual or automatic.

Manual mode : Only manual channel activation (threshold and priority are ignored).

Automatic mode : With priority and threshold channel activation.

- Last Mic. On Designated

  Defines which channel will stay open (Only if "Last Mic. On" is activated).

  the channel will stay open even if it's muted.

- Mute

  Mutes or un-mutes the channel input.

- Invert

  Inverts the polarity of the input channel.

- Solo

  Solos the input channel. (the channel goes directly to the output).

- Gate On

  Shows activation status of the input channel.

- Level

  Adjusts the level of the output channel.

- Manual mode

  Only used in manual mode (see "Manual/Automatic"), It allows manual channel activation.

- Automatic mode priority level

  Sets the priority level of the channel. 1 is the most priority.

- Output Mute

  Mutes or un-mutes the module's output.

- Output Level

  Sets the module's output level.

- Threshold

  Sets the threshold for channel activation.

- Background Threshold

  Sets how the input signal is interpreted to open a gate. The background level will increase the input activation level.

- Hold Time

  Sets the threshold for channel activation.

- Attenuation Response Time

  Sets a pre delay time for automatic attenuation. That's means the transition time of the attenuation when a channel is added or removed.

- NOM Attenuation Step

  Sets the amount of attenuation to be applied each time the number of activated channels increases.

- NOM Attenuation Max

  Limits the amount of attenuation to be applied.

- Attenuation Gain

  Indicates the amount of current gain reduction.

- Open Mic. Limits

  Chooses the maximum number of simultaneously activated channels.

- NOM Attenuation Mode

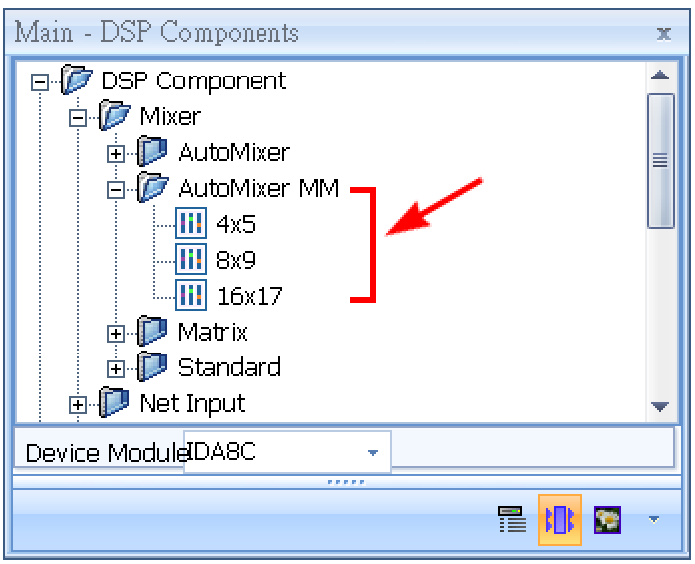
  Chose here if you want a linear attenuation or a logarithmic attenuation which is more near of the reality.

  Eq1: (NOM-1) x step (same attenuation for each addicting channel).

  Eq2: log2(NOM) x step.

6.15.2 AutoMixer MM

❖Component Template

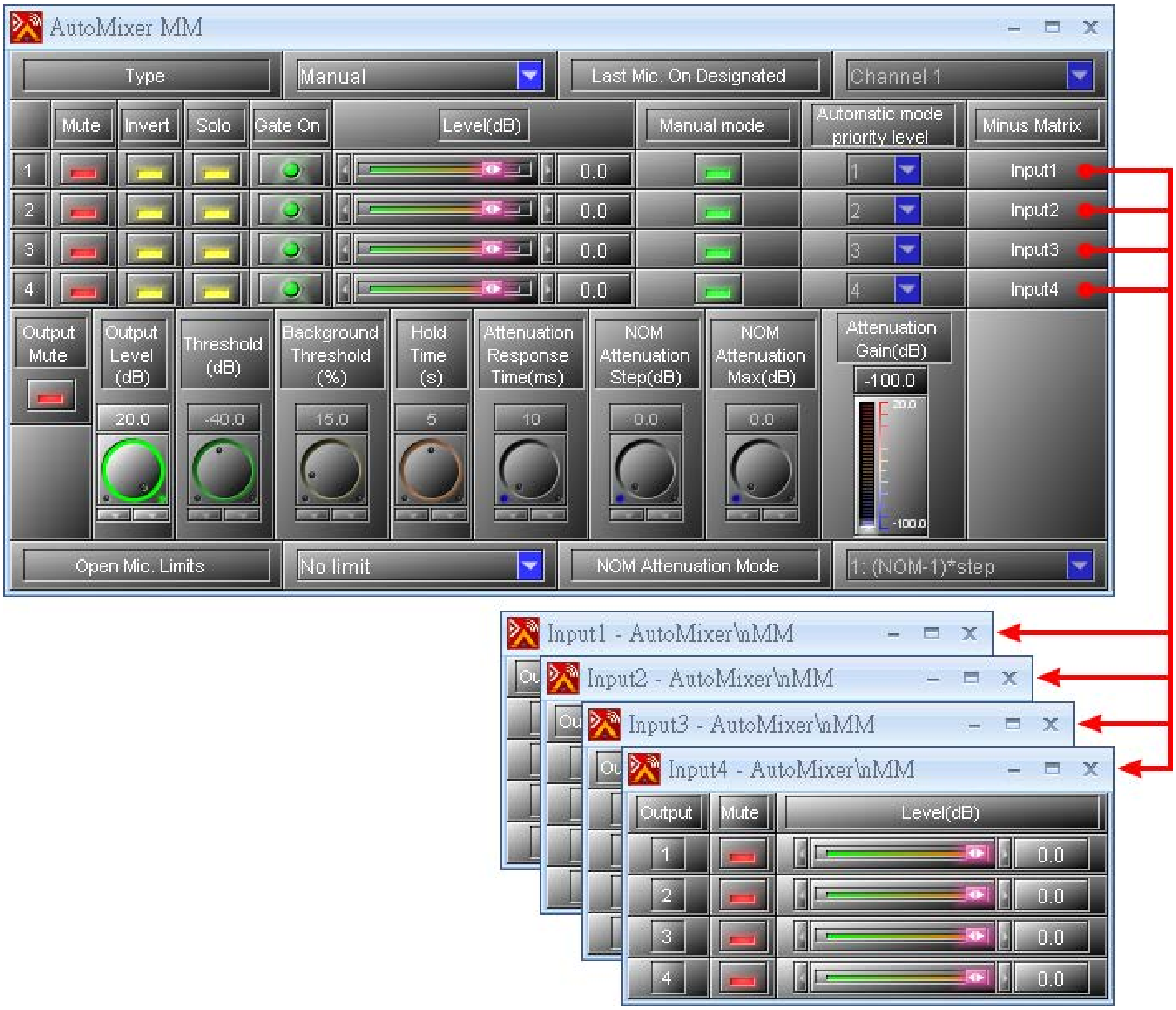


❖Component Appearance

❖Description

This component Allows mix several signals to one output. The Activation of channel(s) can be done manually or automatically by signal threshold and priority. The output 1 is the standard automixer output like you have on the auto-mixer component. In more, on each input channels, you can open a mini-mixer allowing you to send the input signal in different outputs with different level of mixing.

❖ Control Window



❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Type	Manual	-	-	-	-
Last Mic. On Designated	Channel 1	-	-	-	-
Mute	On	-	-	-	-
Invert	Off	-	-	-	-
Solo	Off	-	-	-	-
Gate On	Off	-	-	-	-
Input Level	0	-90.0	20.0	0.1	dB
Manual mode	On	-	-	-	-
Automatic mode priority level	1	2	3	4	-
Output Mute	Off	-	-	-	-

Name	Initial	Minimum	Maximum	Precision	Unit
Matrix Output Mute	Off	-	-	-	-
Matrix Output Level	0	-90.0	0	0.1	dB
Output Level	0	-90.0	20.0	0.1	dB
Threshold	-40.0	-90.0	20.0	0.1	dB
Background Threshold	15.0	0	100.0	0.1	dB
Hold Time	5	0	10.0	0.1	Second
Attenuation Response Time	10	10	10000	1	ms
NOM Attenuation Step	0	0	6	0.1	dB
NOM Attenuation Max	0	0	100	0.1	dB
Attenuation Gain	-100.0	20.0	-100.0	0.1	dB
Open Mic. Limits	No limit	-	-	-	-
NOM Attenuation Mode	1: (NOM-1)*step	-	-	-	-

❖Element Description

- Type

  Select the mode, manual or automatic.

  Manual mode : Only manual channel activation (threshold and priority are ignored).

  Automatic mode : With priority and threshold channel activation.

- Last Mic. On Designated

  Defines which channel will stay open (Only if "Last Mic. On" is activated).

  ⚠ the channel will stay open even if it's muted.

- Mute

  Mutes or un-mutes the channel input.

- Invert

  Inverts the polarity of the input channel.

- Solo

  Solos the input channel. (the channel goes directly to the output).

- Gate On

  Shows activation status of the input channel.

- Level

  Adjusts the level of the output channel.

- Manual mode

  Only used in manual mode (see "Manual/Automatic"), It allows manual channel activation.

- Automatic mode priority level

  Sets the priority level of the channel. 1 is the most priority.

- Output Mute

  Mutes or un-mutes the module's output.

- Matrix Output Mute

  Mute the input for the output mixing. This is different with the Mute element, this element only affect the output channel itself.

- Matrix Output Level

  The attenuation of input for the output mixing.. This is different with the Mute element, this element only affect the output channel itself.

- Output Level

  Sets the module's output level.

- Threshold

  Sets the threshold for channel activation.

- Background Threshold

  Sets how the input signal is interpreted to open a gate. The background level will increase the input activation level.

- Hold Time

  Sets the threshold for channel activation.

- Attenuation Response Time

  Sets a pre delay time for automatic attenuation. That's means the transition time of the attenuation when a channel is added or removed.

- NOM Attenuation Step

  Sets the amount of attenuation to be applied each time the number of activated channels increases.

- NOM Attenuation Max

  Limits the amount of attenuation to be applied.

- Attenuation Gain

  Indicates the amount of current gain reduction.

- Open Mic. Limits

  Chooses the maximum number of simultaneously activated channels.

- NOM Attenuation Mode

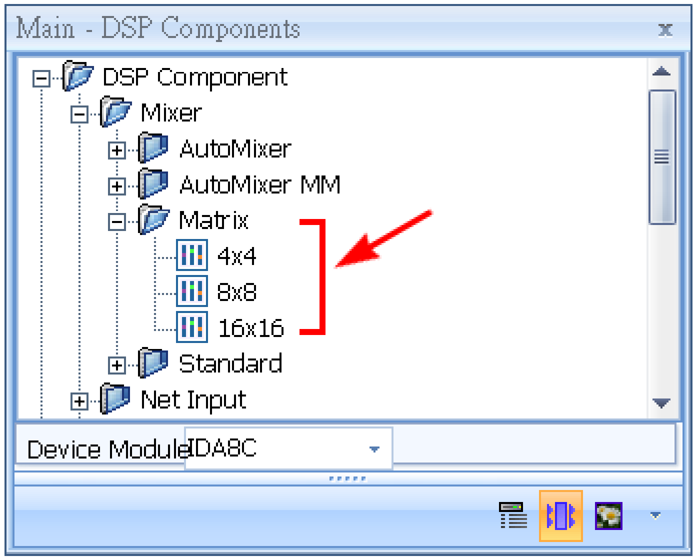
  Chose here if you want a linear attenuation or a logarithmic attenuation which is more near of the reality.

  Eq1: (NOM-1) x step (same attenuation for each addicting channel).

  Eq2: log2(NOM) x step.

6.15.3 Matrix

❖ Component Template



❖ Component Appearance

❖ Description

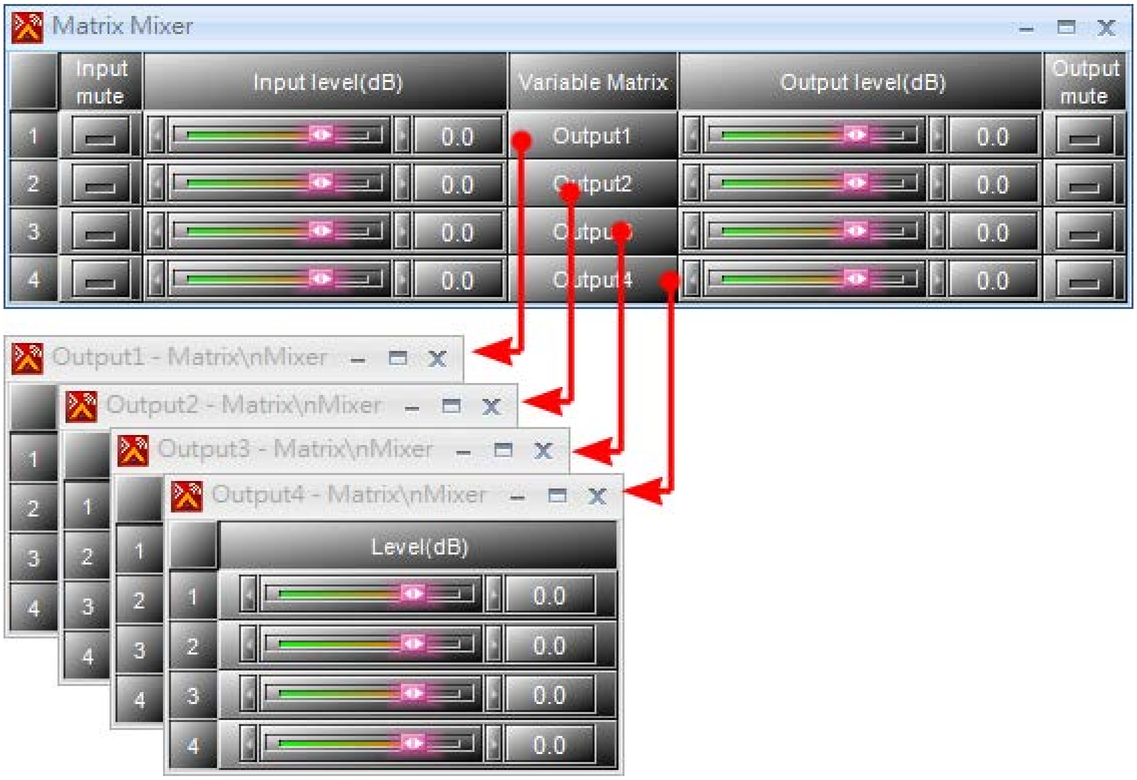
Matrix Mixers offer three main functionalities:

- Mixing: summing signals together..
- Routing: allows the routing of source signals to external processing components.

Clicking the orange colored function [Output(n)] buttons under the label 'Variable Matrix' in the middle of the window opens the sub control windows of the particular output channels.

In here you can create an individual mix of the input signals independently for the specified output. By default all levels are set to 0 db, which basically means, that all audio signals at the input are mixed with no attenuation and appear like this on every output.

❖ Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Input Mute	Off	-	-	-	-
Input Level	0	-60.0	20.0	0.1	dB
Output Level	0	-60.0	20.0	0.1	dB
Output Mute	Off	-	-	-	-
Matrix Output Level	0	-60.0	20.0	0.1	dB

❖Element Description

- Input Mute

  Mutes or un-mutes the channel input.

- Input Level

  Mutes or un-mutes the channel input.

- Output Level

  Sets the master output level of the channel.

- Output Mute

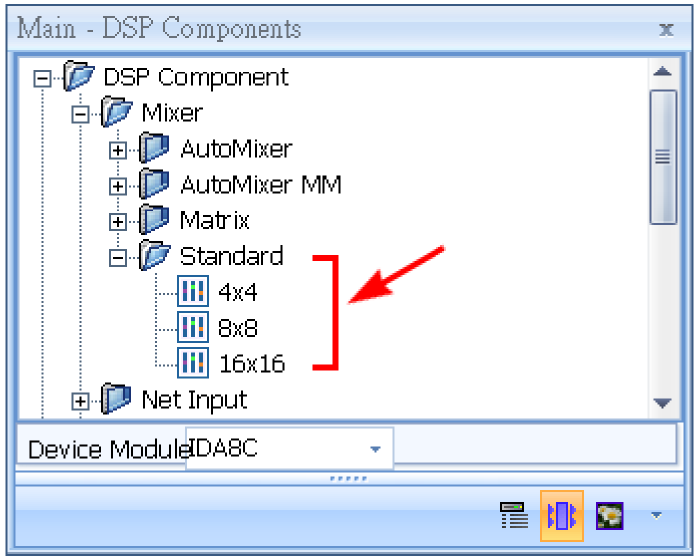
  Mutes or un-mutes the channel output.

- Matrix Output Level

  Set the input level on the particular output channel for mixing.

6.15.4 Standard

❖ Component Template



❖ Component Appearance

❖ Description

The matrix module is similar to the standard mixer, with the main difference, that there is no ability to make independent mixes for each output channel. So the module primary is used for simple routing purposes.

❖ Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Input Mute	Off	-	-	-	-
Input Level	0	-90.0	20.0	0.1	dB
Output Mute	Off	-	-	-	-
Output Level	0	-90.0	20.0	0.1	dB
Routing	*1	-	-	-	-

*1: The initial value of Routing element is on for the [Input 1:Output 1], [Input 2:Output 2], ..., [Input N:Output N], the others are all with initial value off.

❖Element Description

- Input Mute

  Mutes or un-mutes the channel input.

- Input Level

  Sets the input level of the channel.

- Output Mute

  Mutes or un-mutes the channel output.

- Output Level

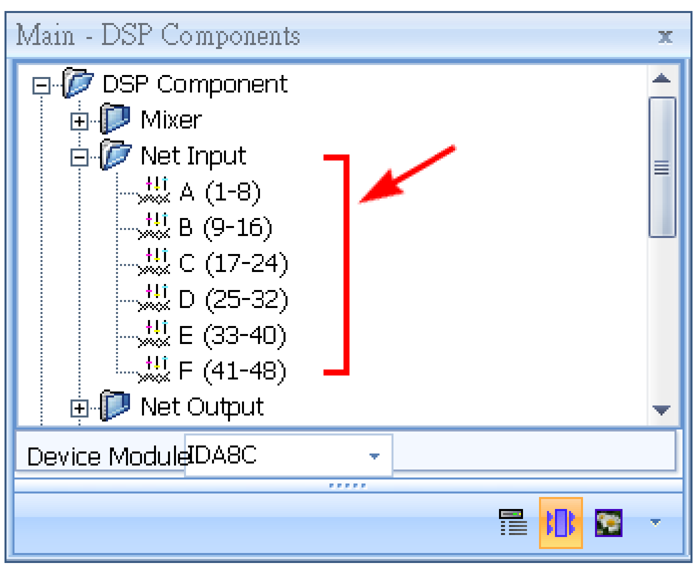
  Sets the master output level of the channel.

- Routing

  Turns on/off the input on the particular output channel.

6.16 Net Input

❖Component Template



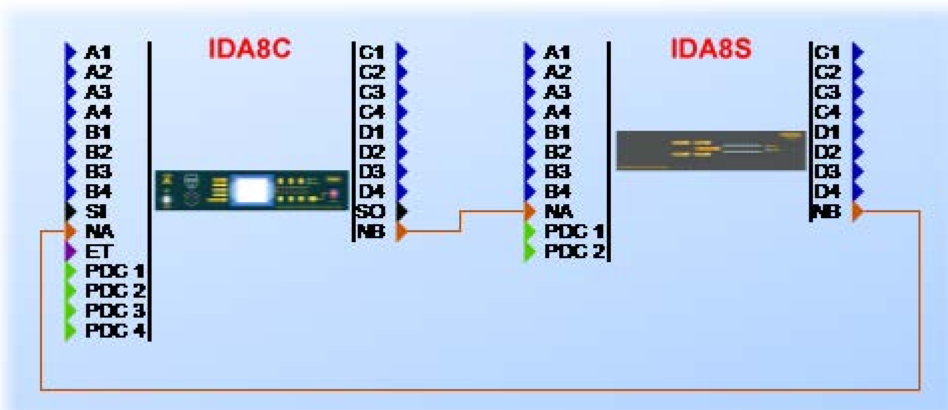
❖Component Appearance

❖ Description

This component transfer audio signals over Ateis Net.The signal goes into a channel of Net Output component and is able to retrieve from output with the same channel or any Net Input component, i. e. the signal path is one to many relationship, There are 48 channels available to carry signals, they are numbered and identified by the number. For example, a signal goes into channel 1 of component Net Output A, you can get it from the component Net Input A channel 1.

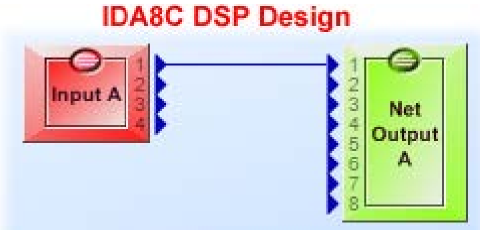
The following article is a simple example using IDA8 devices to show the configuration of Net Input and Net Output.

In Device editing window, create an IDA8C and an IDA8SAB, wires connect NB, NA pins on the block to make up a Ateis Net system:

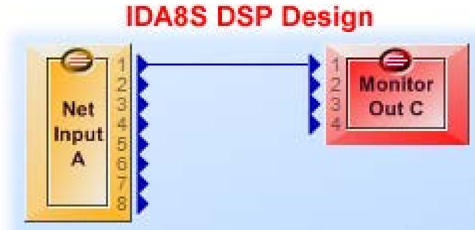


In DSP design window of IDA8C, it utilizes the Net Input/Output channel 1 to transport audio signal from Input A channel in IDA8C to Monitor Out C channel 1:

IDA8C DSP Design

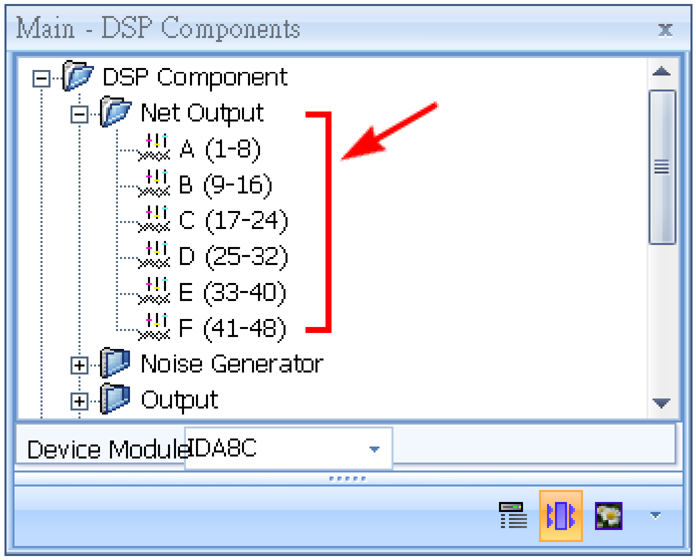


IDA8S DSP Design

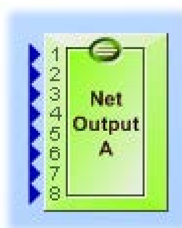


6.17 Net Output

❖Component Template



❖Component Appearance



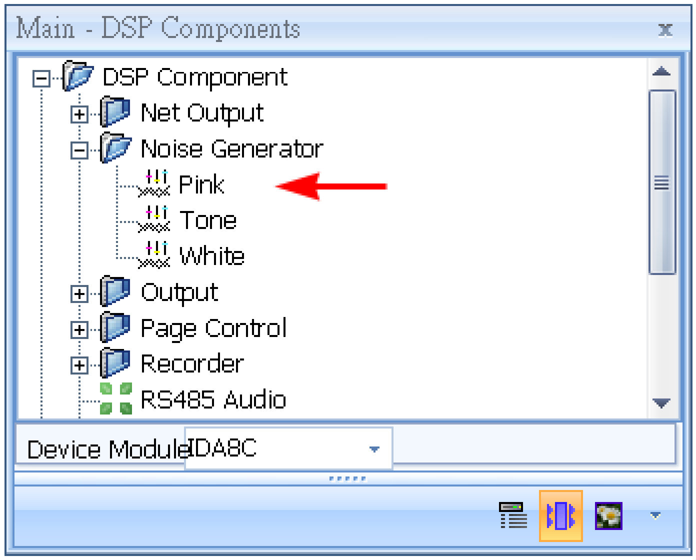
❖Description

This component transfer audio signals over Ateis Net. It must work with Net Input, please go to the topic Net Input for more details.

6.18 Noise Generator

6.18.1 Pink

❖ Component Template



❖ Component Appearance

❖ Description

Pink noise or 1/*f* noise is a signal or process with a frequency spectrum such that the power spectral density is inversely proportional to the frequency. In pink noise, each octave carries an equal amount of noise power.

❖ Control Window

❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	-	-	-	-	-
Level	-100.0	-100.0	7.0	0.1	dB

❖ Element Description

- Mute

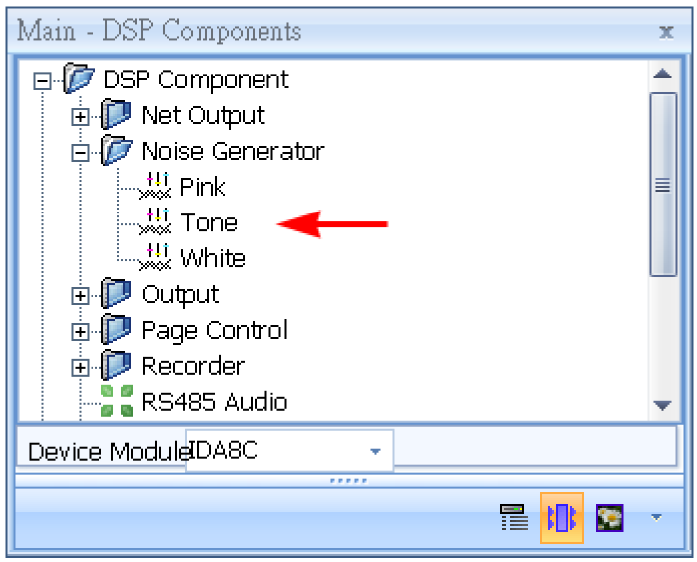
    Mute the sound coming out from the component.

- Level

    Level of the pink noise.

6.18.2 Tone

❖Component Template



❖Component Appearance

❖Description

Generates either tone (single frequency) or sweep sequence (between two fixed frequencies and in a defined interval of time).

❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Level	-100.0	-100.0	20.0	0.1	dB
Tone freq.	1000	20	20000	1	Hz
Start freq.	1000	20	20000	1	Hz
Stop freq.	1000	20	20000	1	Hz
Inc. Time	500	100	60000	1	ms
Mute	-	-	-	-	-
Sweep	-	-	-	-	-
Tone	-	-	-	-	-
Sweep Type	-	-	-	-	-

❖Element Description

- Level

  Level of the signal.

- Tone freq.

  If Tone mode is activated select the frequency played.

- Start freq.

  If Tone mode is activated select the frequency played.

- Stop freq.

  Select the maximum frequency in sweep mode.

- Inc. Time

  Select the time between the frequencies change in sweep mode.

- Mute

  Mute the sound coming out from the component.

- Sweep

  Active the sweep mode (sweep of frequency).

- Tone

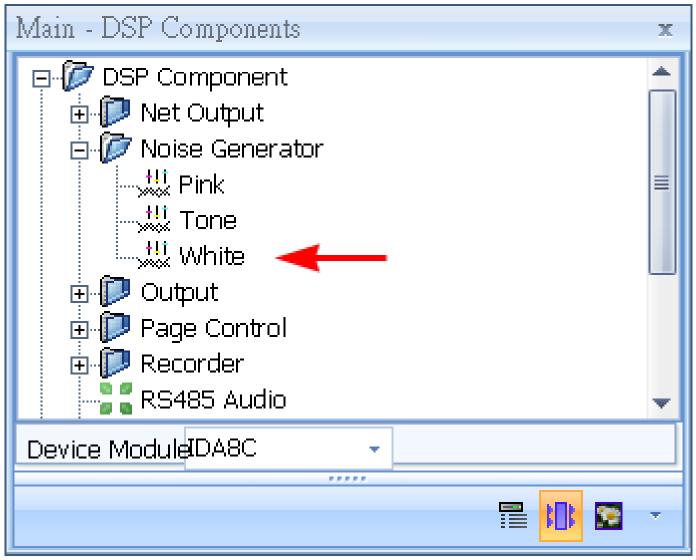
  Active Tone mode (one single frequency played).

- Sweep Type

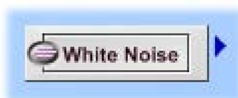
  Choose the interval between Min and Max frequencies (1Octave, 1/3 Octave, ..., etc.).

6.18.3 White

❖ Component Template



❖ Component Appearance



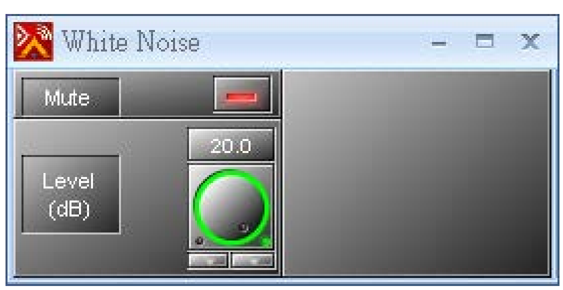
❖ Description

This component generates a White noise. The White noise is a random signal (or process) with a flat power spectral density. In other words, the signal contains equal power within a fixed bandwidth at any center frequency.

An infinite-bandwidth, white noise signal is purely a theoretical construction. By having power at all frequencies, the total power of such a signal is infinite and therefore impossible to generate. In practice, however, a signal can be "white" with a flat spectrum over a defined frequency band.

As example, it can be used to set up the equalization for a concert or other performance in a venue, a short burst of white or pink noise is sent through the PA system and monitored from various points in the venue so that the engineer can tell if the acoustics of the building naturally boost or cut any frequencies. The engineer can then adjust the overall EQ to ensure a balanced mix.

❖ Control Window



❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Mute	-	-	-	-	-
Level	-100.0	-100.0	20.0	0.1	dB

❖Element Description

- Mute

  Mute the sound coming out from the component.

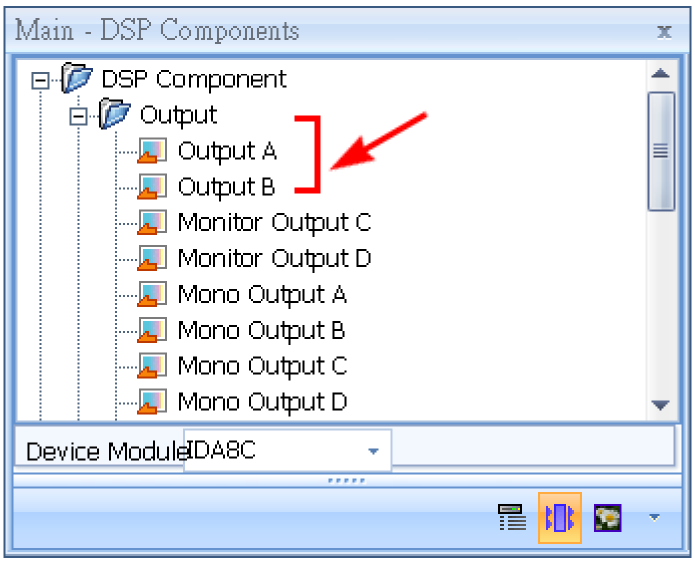
- Level

  Level of the pink noise.

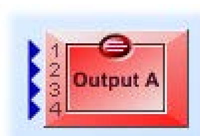
6.19 Output

6.19.1 Output

❖Component Template



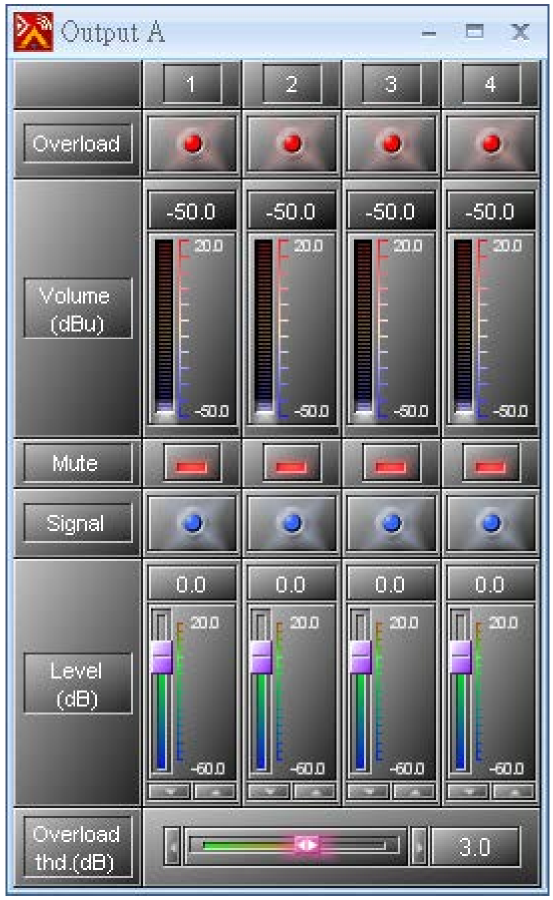
❖Component Appearance



❖Description

The analogue output modules represent the physical analogue outputs accessible on the rear panel of the device. There are several separate modules for use with each of the audio board. Analogue input and output modules are added automatically to the DSP window of the relevant processor by defining their module configuration at the time you create them and should not be altered.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	0.1	dBu
Mute	Off	-	-	-	-
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload thd.	3.0	-20.0	20.0	0.1	dB

❖Element Description

- Overload

  Indicates channel level above chosen Overload Threshold (dB).

- Volume

  Meter showing the channel level.

- Mute

  Mutes or un-mutes the output channel; LED, lit red with channel muted.

- Signal

  To indicate the channel is active or not.

- Level

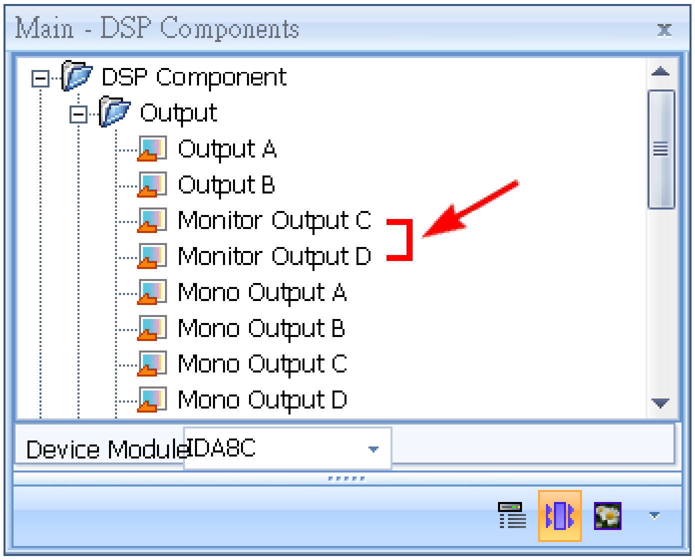
  Signal output level of the module.

- Overload thd.

  Sets the threshold of the overload indicator.

6.19.2 Monitor Output

❖Component Template

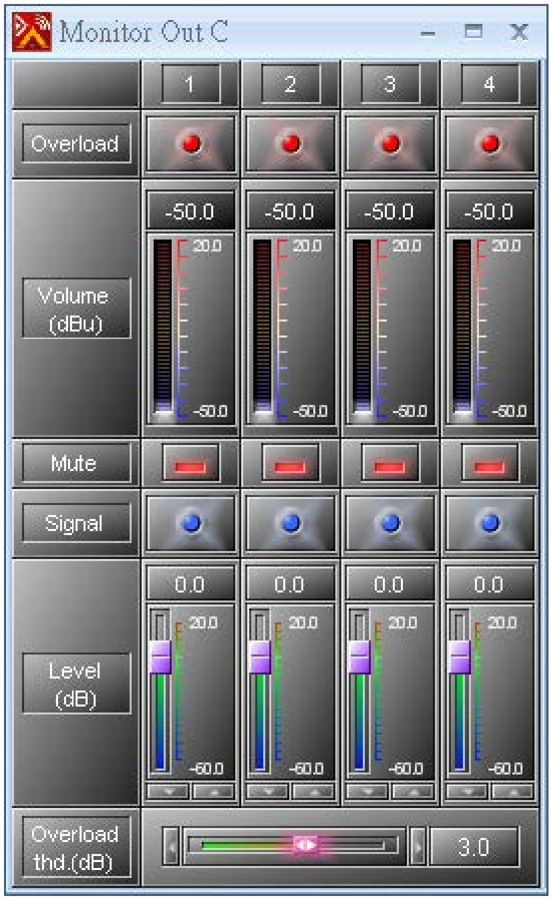


❖Component Appearance

❖Description

The monitor output modules represent the amplifier zone output accessible on the rear panel of the device. There are 2 separate modules for use with each of the zone board 1, 2. Analogue input and output modules are added automatically to the DSP window of the relevant processor by defining their module configuration at the time you create them and should not be altered.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	0.1	dBu
Mute	Off	-	-	-	-
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload thd.	3.0	-20.0	20.0	0.1	dB

❖Element Description

- Overload

  Indicates channel level above chosen Overload Threshold (dB).

- Volume

  Meter showing the channel level.

- Mute

  Mutes or un-mutes the output channel; LED, lit red with channel muted.

- Signal

  To indicate the channel is active or not.

- Level

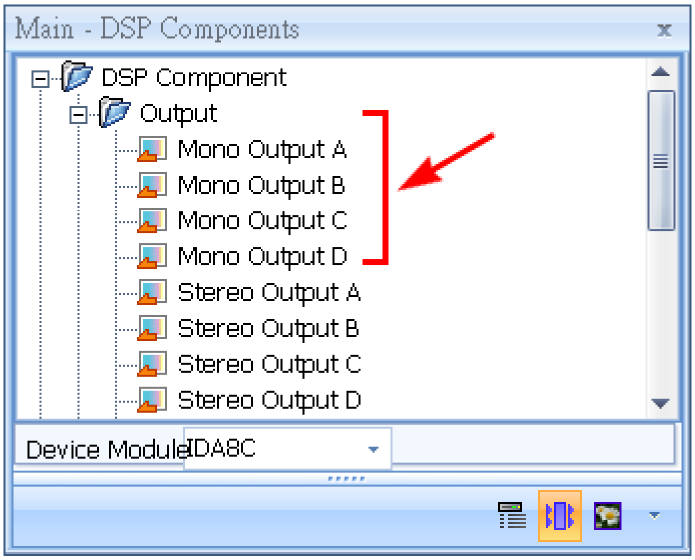
  Signal output level of the module.

- Overload thd.

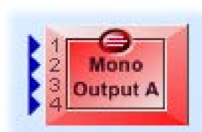
  Sets the threshold of the overload indicator.

6.19.3 Mono Output

❖ Component Template



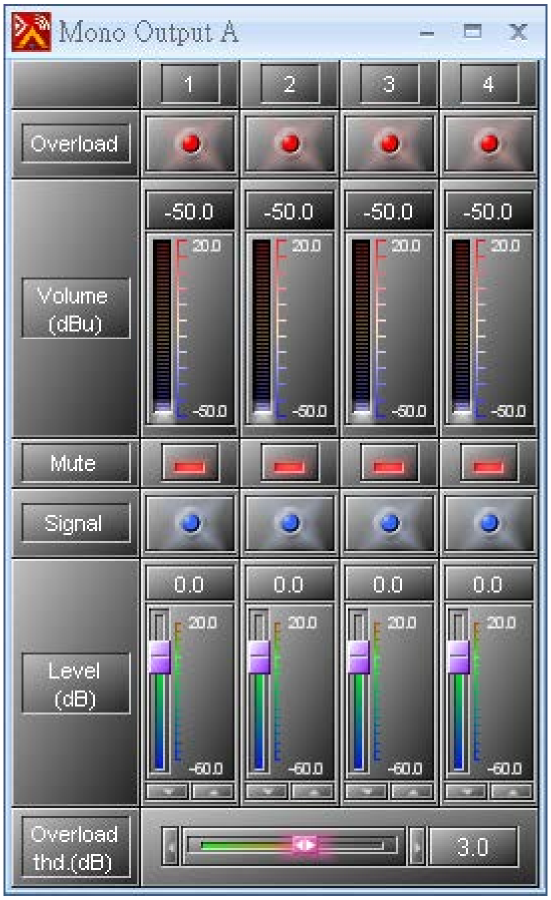
❖ Component Appearance



❖ Description

For AES/EBU output, the mono output modules represent the physical outputs accessible on the rear panel of the device. There are several separate modules for use with each of the audio board. Input and output modules are added automatically to the DSP window of the relevant processor by defining their module configuration at the time you create them and should not be altered.

❖ Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	0.1	dBu
Mute	Off	-	-	-	-
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload thd.	3.0	-20.0	20.0	0.1	dB

❖Element Description

- Overload

  Indicates channel level above chosen Overload Threshold (dB).

- Volume

  Meter showing the channel level.

- Mute

  Mutes or un-mutes the output channel; LED, lit red with channel muted.

- Signal

  To indicate the channel is active or not.

- Level

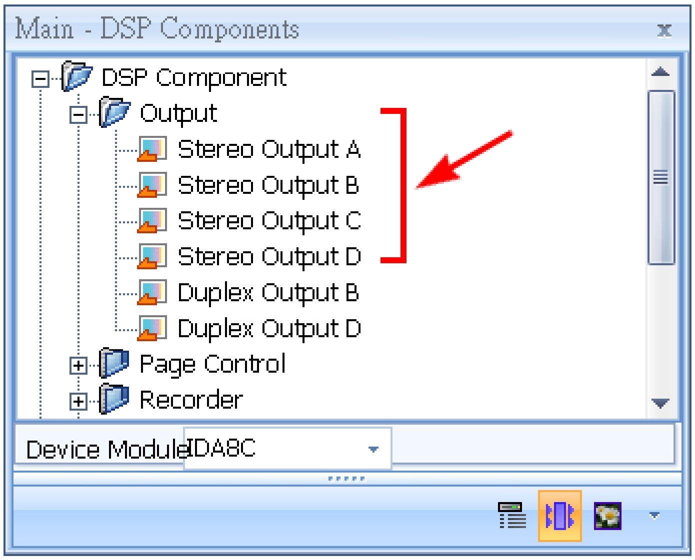
  Signal output level of the module.

- Overload thd.

  Sets the threshold of the overload indicator.

6.19.4 Stereo Output

❖Component Template

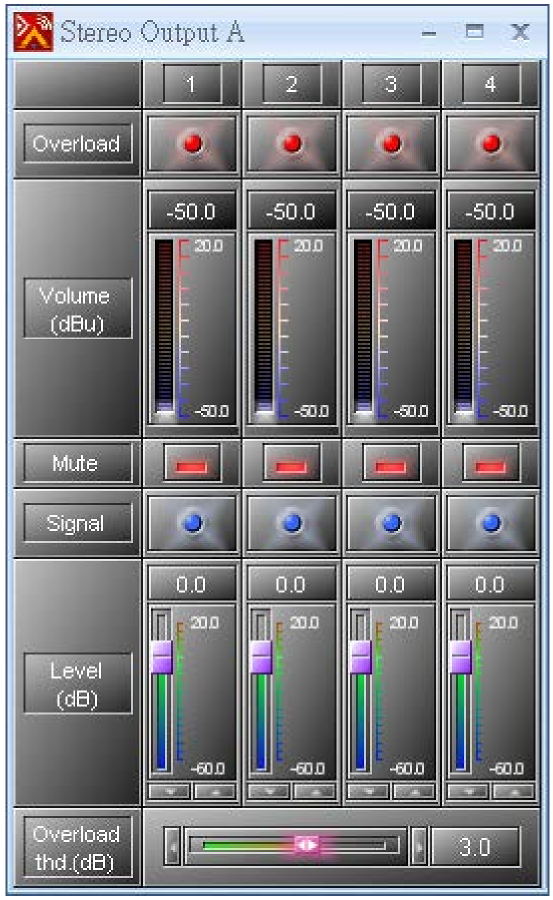


❖Component Appearance

❖Description

For AES/EBU output, the stereo output modules represent the outputs accessible on the rear panel of the device. There are several separate modules for use with each of the audio board. Input and output modules are added automatically to the DSP window of the relevant processor by defining their module configuration at the time you create them and should not be altered.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	0.1	dBu
Mute	Off	-	-	-	-
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload thd.	3.0	-20.0	20.0	0.1	dB

❖Element Description

- Overload

  Indicates channel level above chosen Overload Threshold (dB).

- Volume

  Meter showing the channel level.

- Mute

  Mutes or un-mutes the output channel; LED, lit red with channel muted.

- Signal

  To indicate the channel is active or not.

- Level

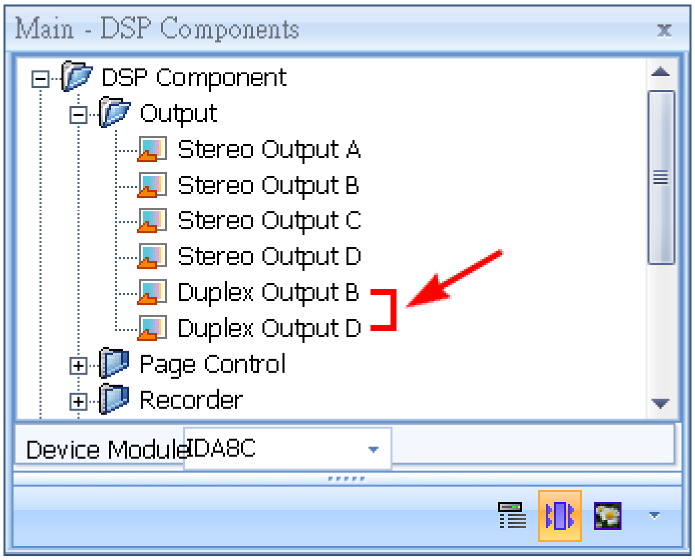
  Signal output level of the module.

- Overload thd.

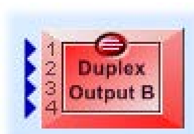
  Sets the threshold of the overload indicator.

6.19.5 Duplex Output

❖Component Template



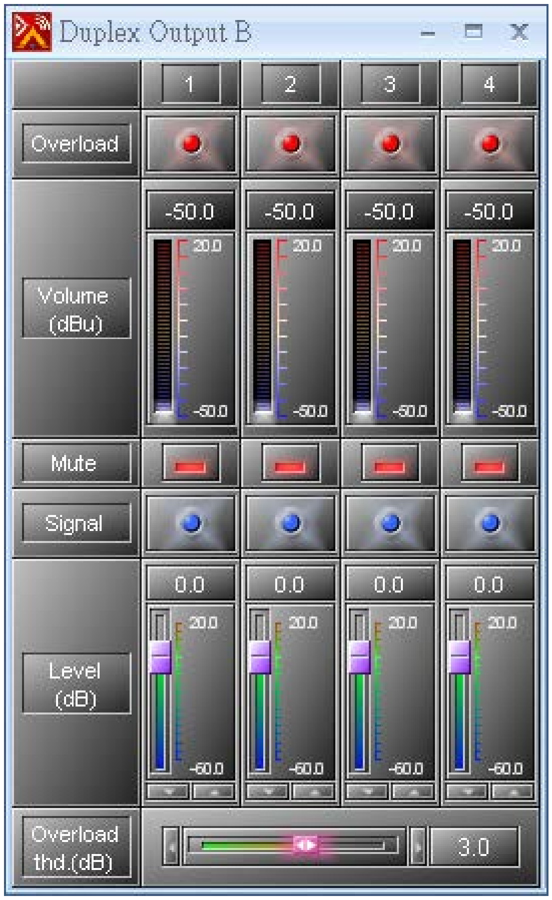
❖Component Appearance



❖Description

For AES/EBU output, the duplex output modules represent the physical outputs accessible on the rear panel of the device. There are several separate modules for use with each of the audio board. Input and output modules are added automatically to the DSP window of the relevant processor by defining their module configuration at the time you create them and should not be altered.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Overload	Off	-	-	-	-
Volume	-50.0	-50.0	20.0	0.1	dBu
Mute	Off	-	-	-	-
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload thd.	3.0	-20.0	20.0	0.1	dB

❖Element Description

- Overload

  Indicates channel level above chosen Overload Threshold (dB).

- Volume

  Meter showing the channel level.

- Mute

  Mutes or un-mutes the output channel; LED, lit red with channel muted.

- Signal

  To indicate the channel is active or not.

- Level

  Signal output level of the module.

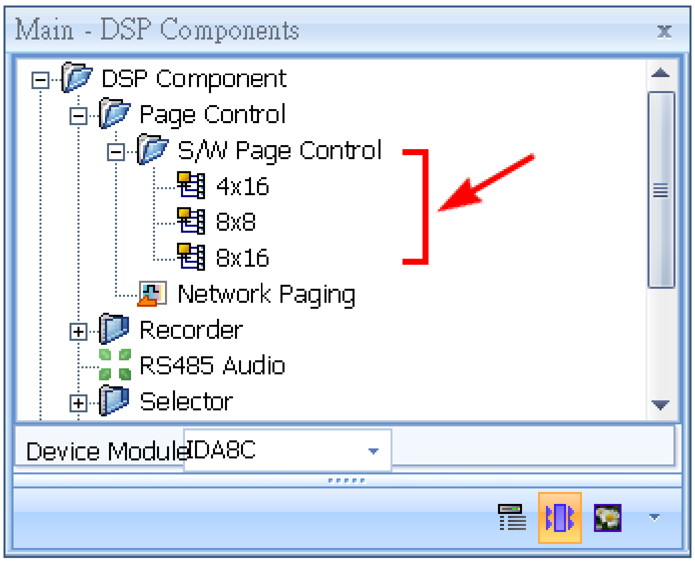
- Overload thd.

  Sets the threshold of the overload indicator.

6.20 Page Control

6.20.1 S/W Page Control

❖Component Template



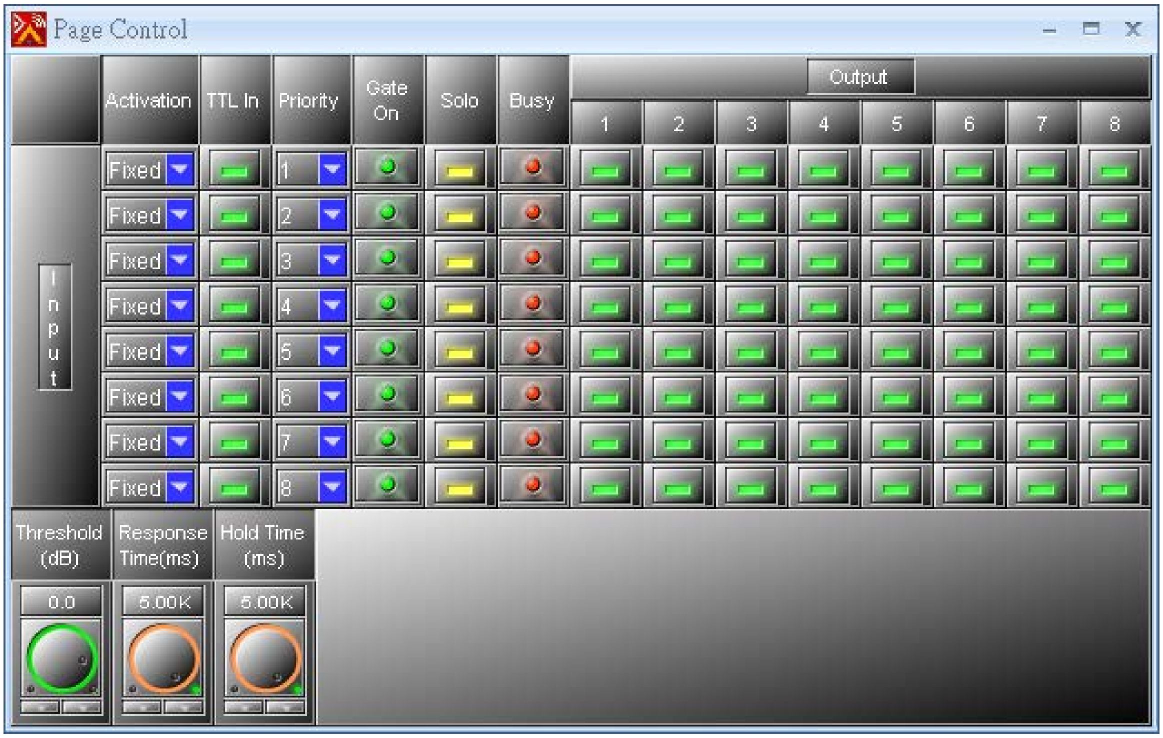
❖Component Appearance

❖Description

With this component you can active and route the audio inputs to the outputs. Caution, this is not a mixer, only one input can be routed to an output.

You can never have two inputs channels mixed to an output (even if they have the same priority), in this case the busy led of the lower channel number will light and theses channels won't be routed. Each outputs work separately. On each one output, only the channels with the gate ON and with the highest priority will be routed.

❖Control Window



❖ Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Activation	Fixed	-	-	-	-
TTL In	Off	-	-	-	-
Priority	*1	-	-	-	-
Gate On	Off	-	-	-	-
Solo	Off	-	-	-	-
Busy	Off	-	-	-	-
Output	Off	-	-	-	-
Threshold	0	-90.0	20.0	0.1	dB
Response Time	10	10	5000	10	ms
Hold Time	3000	10	5000	10	ms

*1: Ch1 = 1, Ch2 = 2, ..., etc.

❖ Element Description

- Activation

  Type of activation of the routing (fixe, with TTL, or with a gate).

- TTL In

  If you want to active the routing with a logical input (rear panel of UAP).

- Priority

  Choose the priority of this input (1 is the biggest priority).

- Gate On

Flash when the gate is active (open).

- Solo

  Route this input.

- Busy

  Lit if the gate of the channel is open, but another channel with highest priority is already activate, and unable this channel to be routed to the output.

  Busy means: gate open, but channel not routed.

- Output

  Click on the desired output’s button to rout input to desired outputs.

- Threshold

  Sets the threshold level above which Gate will open automatically route the signal.

- Response Time

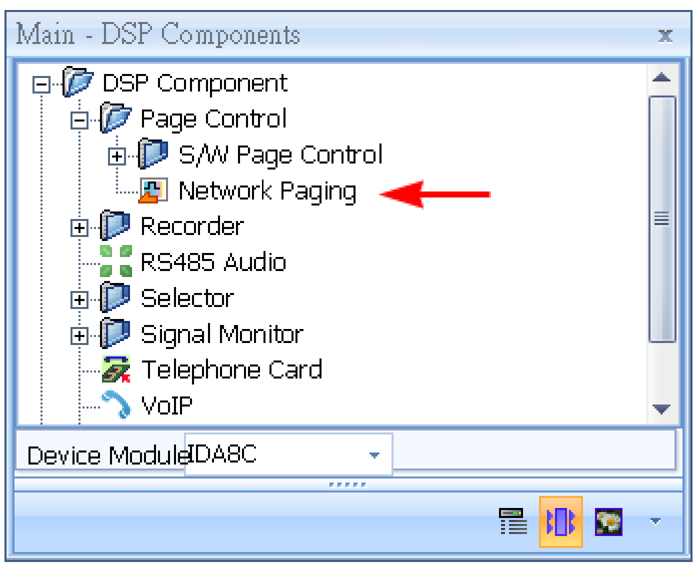
  Adjust the time between the level's detection and the beginning of the Gate operation.

- Hold Time

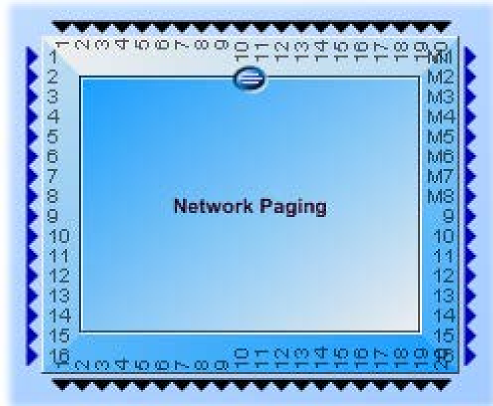
  Adjust the time between the end level's detection and the end of the Gate operation.

6.20.2 Network Paging

❖Component Template



❖Component Appearance

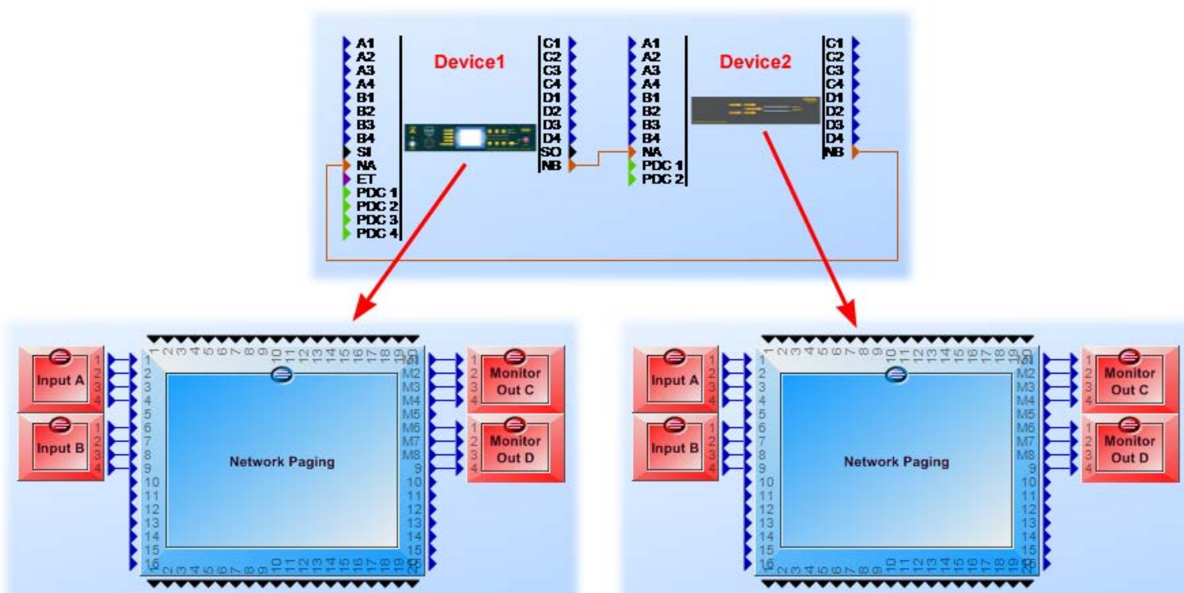


❖Description

The job of Network Paging component is to route input source signal to zones using priority arbitration. The pins on left side represents the paging sources, and the pins on right are zone. Pins on top side are able to controls paging activity, and pins on bottom output logic signals to tell the status of paging activity. Each source can specify which zones want to paging. when a source is request for paging, not always get success, because the zones required maybe already occupied by another source.

❖Network Cascading Paging

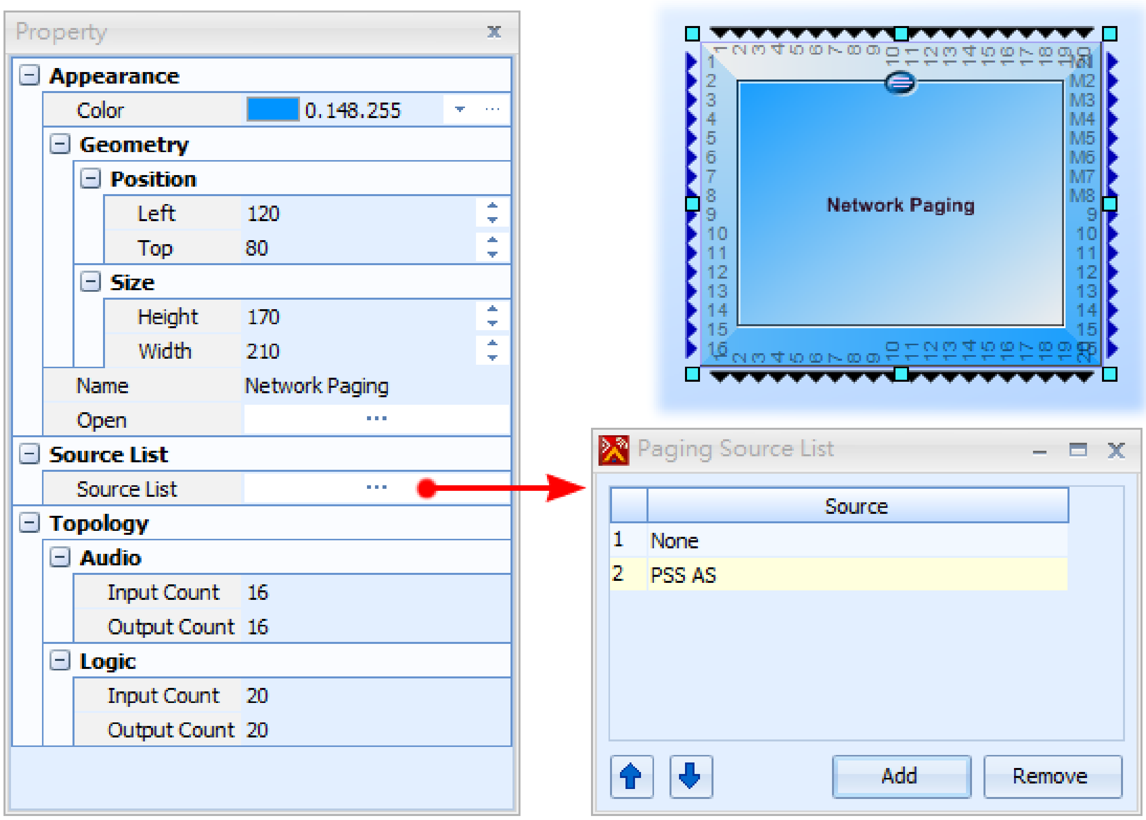
Some application like station PA system, requires a huge number of zones for paging. It need to have multiple Ateis audio processor connected together using Ateis Net, and for each DSP configuration of audio processor, there is a Network Paging component in charge of paging tasks. The following figure is an example to show that:



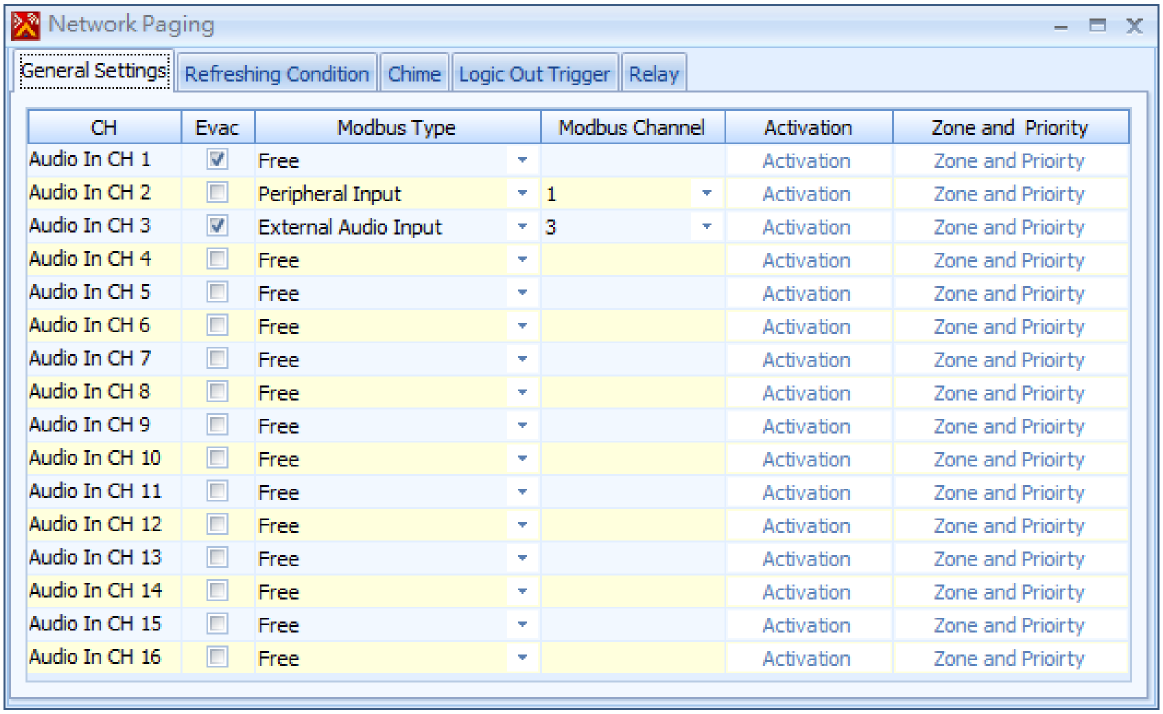
Double click the Network Paging component open the settings window, the following sections describe for parameters for each tab

❖Source Type

A source type is an identification for paging activities. If a zone if occupy by some source for paging, it need to know who did this paging, and who is the owner of the zone. The source type represent it of the case. You can edit source type of Network Paging component by property windows. The procedure is very simple, click Network Paging component on DSP design window, once Network Paging component is focused, the property windows revel all the parameters of it. Click the field of Source List on property window to edit source types.



❖ General Settings



- Evac

  Enable this option to set the paging source is an evacuation source. If an evacuation source is active on paging, A evacuation paging situation is recognized by IDA8 system, and device shows this state on front panel LED GLOBAL EVAC. Also this activity will be logged.

- Modbus Type

  Set the type information of paging source for 3rd party controller which use modbus protocol to get date from Ateis devices.

  Note: This setting is only a label, not affect behavior of Paging Component.

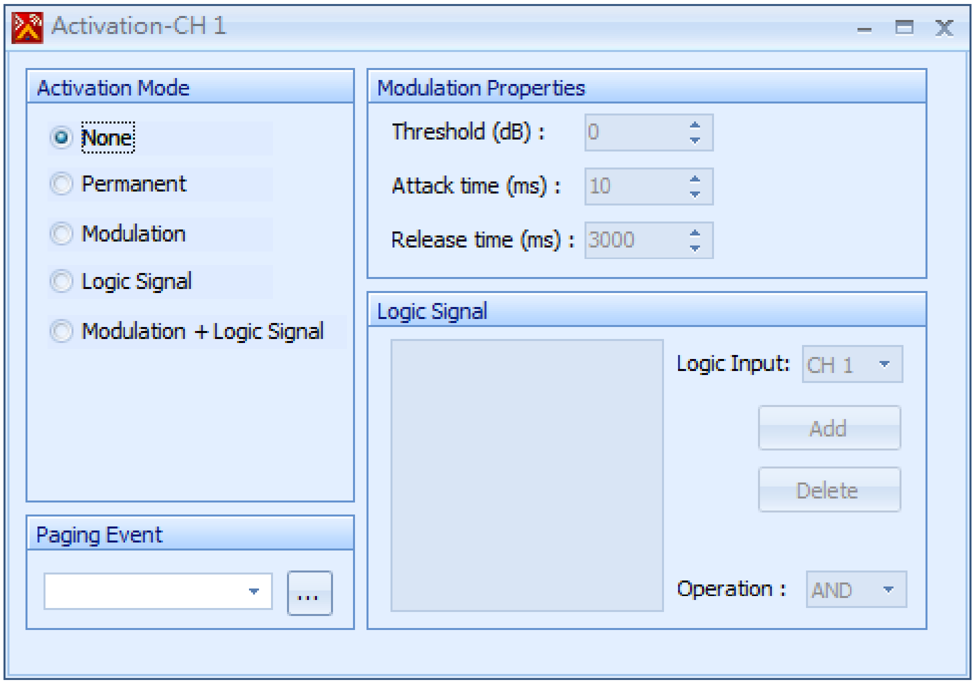
- Modbus Channel

  Set the channel information of paging source for 3rd party controller which use modbus protocol to get date from Ateis devices.

  Note: This setting is only a label, not affect behavior of Paging Component.

- Activation

  Click the button on grid to open window for detail settings:



There are various mode for triggering a paging activity.

○ None

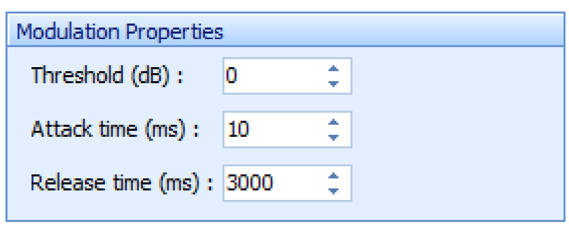
The paging request is not coming from the Network Paging component itself, If you need to do a paging using control console like PPM AS or PSS AS please select this option.

○ Permanent

Means the paging source always request for paging. You can choose this option if the source is background music for keep audio playing for zones continuously.

○ Modulation

Paging request is recognized/released if the conditions on [Modulation Properties] are satisfied.



▪ Threshold / Attack Time

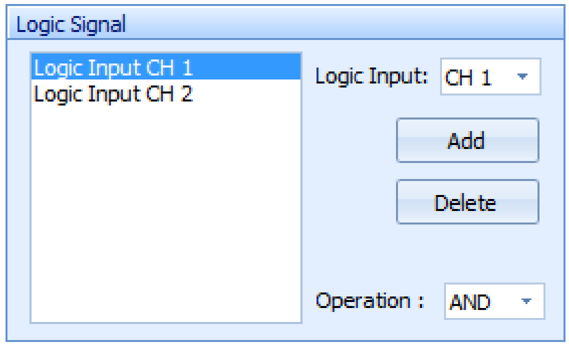
IDA8s acknowledge a paging request of the channel the signal level continuous higher then Threshold over Attack time.

▪ Release Time

IDA8s release the paging request of the channel if the signal level continuous lower than Threshold over Release time.

○ Logic Signal

A paging request of the channel is established depend on the logic signal of input.



Basically if logic from lo to hi, it request for paging, on the other hand form hi to lo for releasing a paging request.

The settings allows you to put multiple channel to do logic operation AND/OR to determine paging activities.

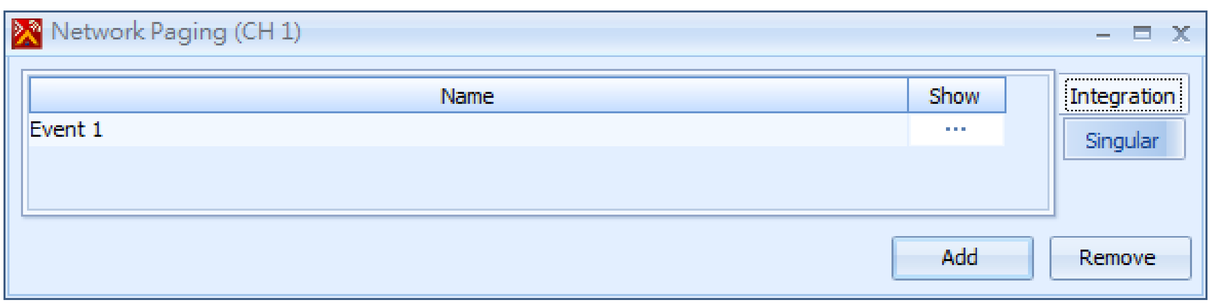
- Modulation + Logic Signal

  Under this mode, IDA8 decide to request a paging if both modulation and logic signal condition are satisfied, but release a paging request if one of condition is unsatisfied.

- Paging Event

  Bind a paging event for zone selection, only events with singular type are selectable. See Zone and Priority section for more details about paging events.

- Zone and Priority



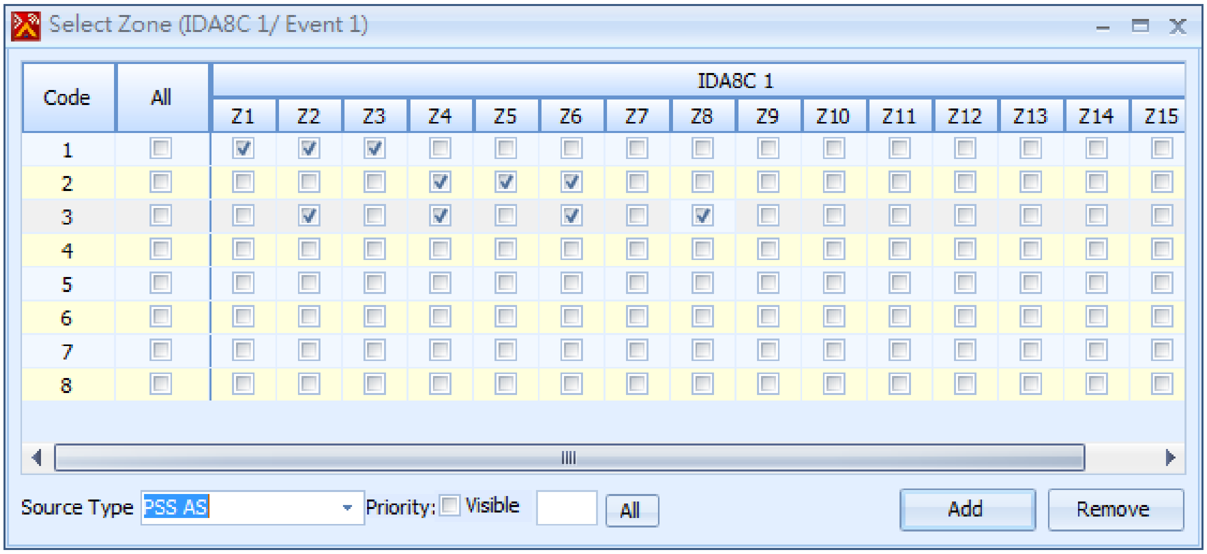
This section mainly talk about the zone section of a paging request, and to specify priority for judgment if there are zones conflict with another paging source which occupy it.

There are two type of paging event:

- Integration

  An integration paging event has multiple code for zone selection, for each code, is able to assign to the key of control console for example PPM AS, PSS AS.

  By clicking the ellipsis button, a settings window of integration paging event is opened:



- Code

  A code is a zone combination so called a selection of zones, the figure above is an example, the zone combination of code is (Z1, Z2, and Z3) means zone 1, 2 and 3 will be selected when requesting a paging.

- All

  To select all zones for the code.

- Zone Selection

  Click the check box to specify a zone is included in a code. The table lists all the zones of paging component under the same Ateis Net.

- Source Type

  Specify the source type to bind with the paging event. When a paging event is active, i.e. paging request is acknowledged by IDA8, you can say the zones are occupied by the audio source bound with that paging event. This info(zone:occupied source) are used in some logic or procedure of paging component implementation for give different results for user. See later sections for more details about how the audio source used to.

- Priority

  There is a priority value for each zone along with code. If there are more than one source attempt to paging the same zone, IDA8 use the priority value to decide which source is allowed to occupy the zone for paging.

  The smaller value with higher priority.

- Priority Visible

  A check box to show or hide the priority setting inside grid.

- Button All

  Set priority value for all zones. The value to apply for all zones is in the editor left to button.

- Button Add

  Add codes to event;

- Button Remove

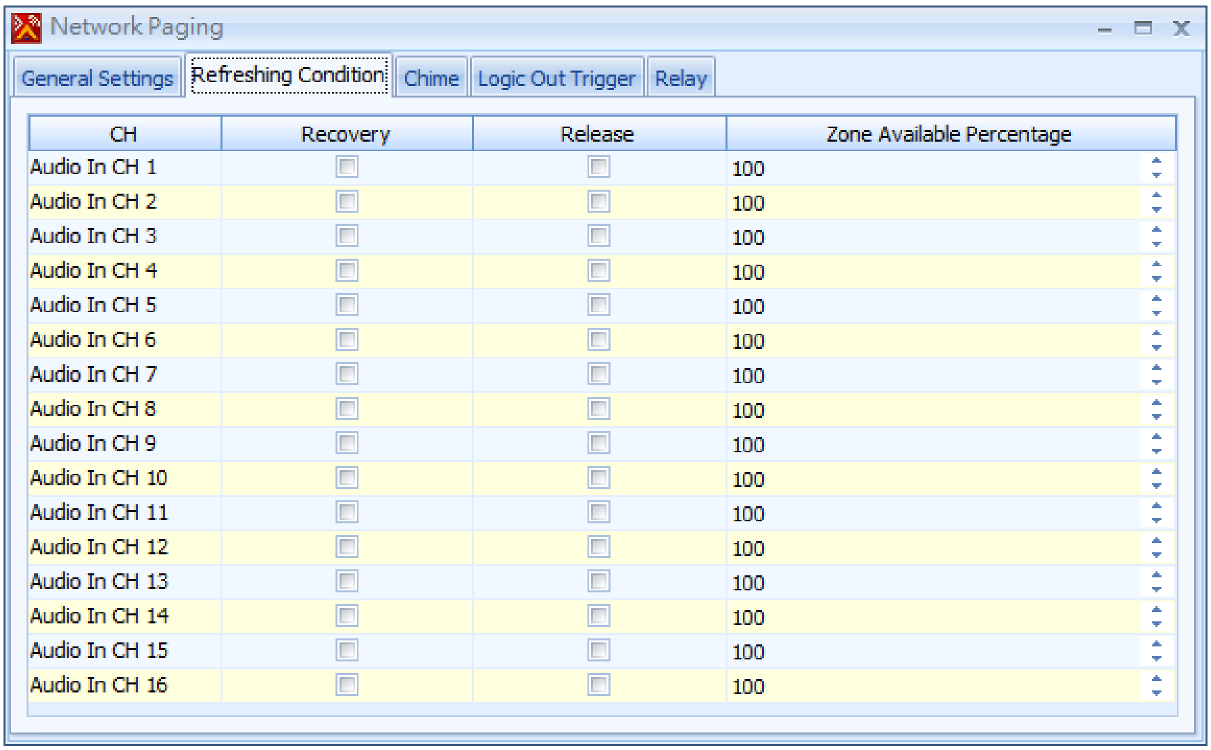
  Remove selected code.

  - Singular

    This event is very similar with integration paging event except the number of code is only one.

❖ Refreshing Condition

This section is talking about the ending behavior of a paging request and the zone available condition.



- Recovery

  If this option is enabled, a paging source will try to take back the zones after another source which occupy the zones finish paging session.

  ⚠ The occupied zone gives back to the active source if Both options Recovery of active source and Release of releasing source are enabled.

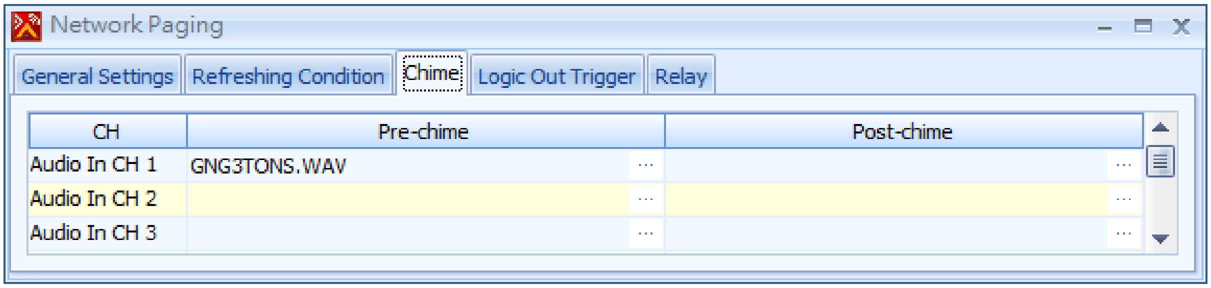
- Release

  If this option is enabled, a paging source will release the zone to other active sources after finish paging session.

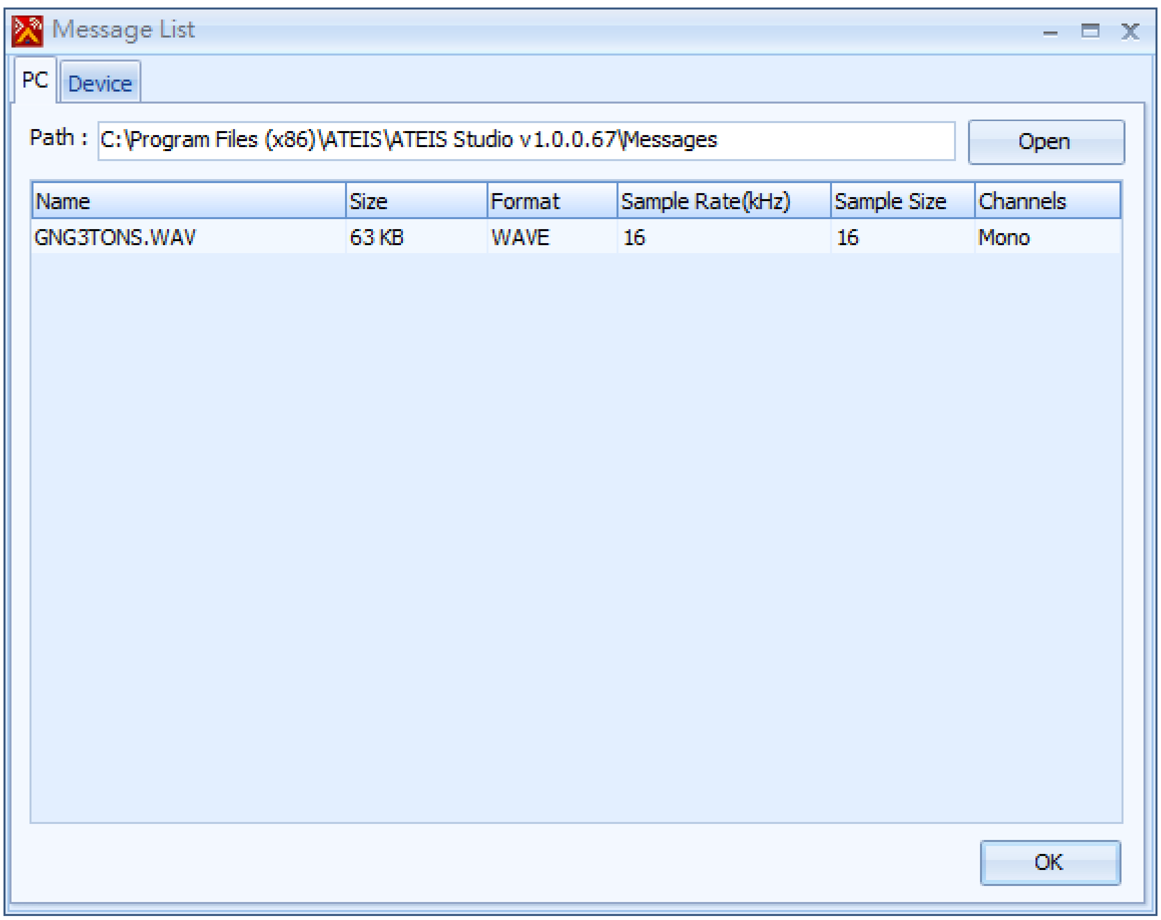
- Zone Available Percentage

This is the condition to determine a paging request can be allowed or denied. If the percentage of available zones is greater than the number on the settings "Zone Available Percentage", the paging is grand. On the contrary, the paging request is denied. The available zones means the zones to be paged are not occupied by other sources or occupied by other sources with lower priority.

❖ Chime

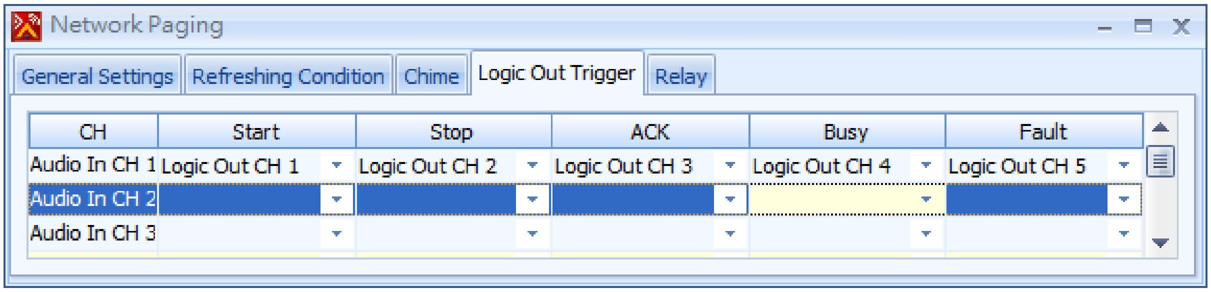


Network Paging component provide the ability of chime playing before and end of an paging session. To specify the chimes binding to a source using the tab [chime] of Network Paging setting window. click the cell on grid, the second layer window opened:



You can select chime listed in grid table of [PC] tab or from [Device] tab. The button [Open] allow you to change the directory for discovery the sound files.

❖Logic Out Trigger



There 20 channel logic output for each Network Paging component, this section shows the configuration about it.

- Start

  Generate a pulse to a specific logic out when paging begins.

- Stop

  Generate a pulse to a specific logic out when paging stoped.

- ACK

  The specific logic output stay hi during paging session.

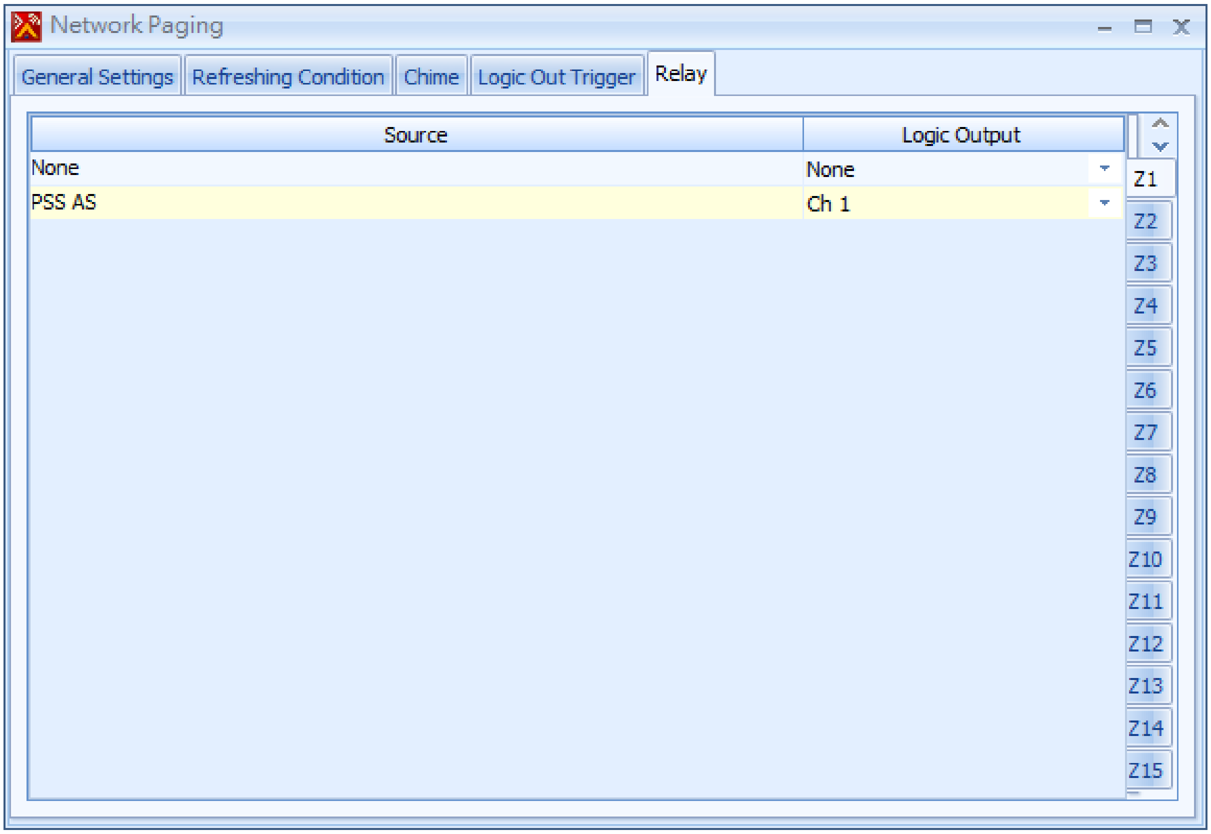
- Busy

  Check any of selected zoned if it is occupied by the other source with higher priority.

- Fault

  The specific logic output stay hi if any selected zone is error.

❖Relay

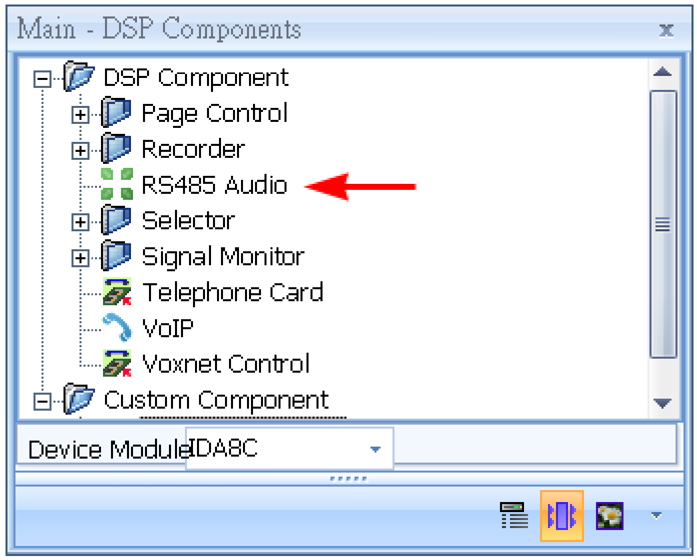


It is able to output a logical hi to a specific output channel in condition of zone occupied by a specific source. In the above figure, logic out channel 1 will stay hi if zone 1 is occupied by source PSS AS.

6.22 RS485 Audio

6.22.1 RS485 Input

❖ Component Template

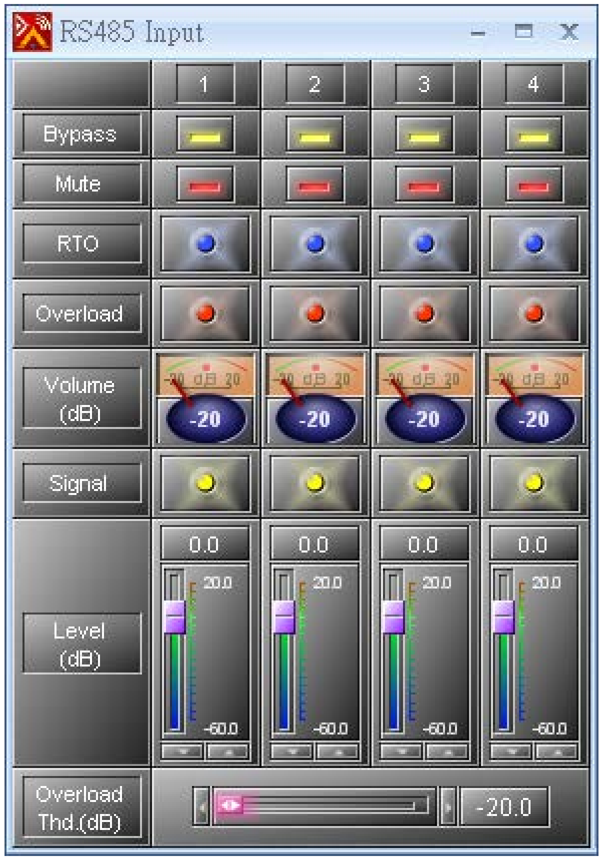


❖ Component Appearance

❖ Description

This component control the RS485 input signals parameters and monitor it. The input component represent the physical analogue inputs accessible on the 485 port of the Audio Processor. There are A~D four ports represent for different channel of 485 port. Some consoles like PSS AS it transfer audio signal via PDC port which is RS485 interface. To process audio signal with other DSP component, you need to create RS485 Input component in design window.

❖ Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Bypass	Off	-	-	-	-
Mute	Off	-	-	-	-
RTO	Off	-	-	-	-
Overload	Off	-	-	-	-
Volume	-20.0	-20.0	20.0	0.1	dB
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload Thd.	-20.0	-20.0	20.0	0.1	dB

❖Element Description

- Bypass

  Bypass audio of the channel.

- Mute

  Mute audio of the channel.

- RTO

  Indicates an input channel 'Routed To Output(s)'.

- Overload

This LED light up if the signal of input channel greater than Overload Thd.

- Volume

  Meter showing the channel RMS level.

- Signal

  Indicates audio signal presence above -30 dB from chosen 'Sensitivity'.

- Level

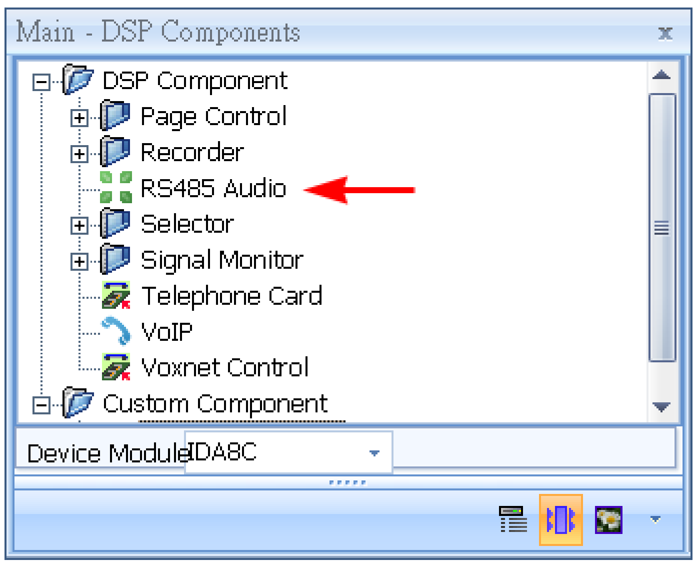
  Signal output level of the channel.

- Overload Thd.

  A threshold value to determine signal of a channel is overload or not.

6.22.2 RS485 Output

❖Component Template

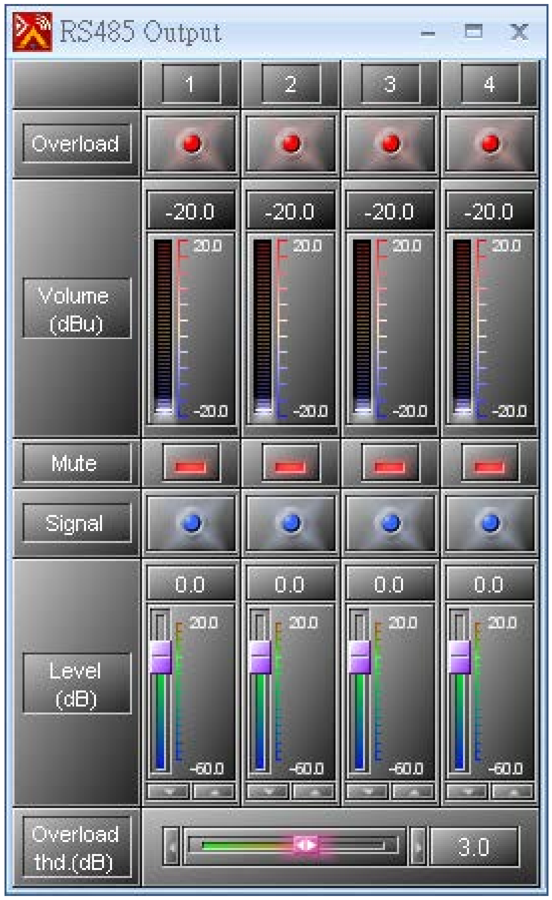


❖Component Appearance

❖Description

This component control the RS485 output signals parameters and monitor it. The output component represent the physical analogue inputs accessible on the 485 port of the Audio Processor. There are A~D four ports represent for different channel of 485 port. Some consoles like PSS AS it transfer audio signal via PDC port which is RS485 interface. To output audio signal via RS485 port, you need to create RS485 output component in design window.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Overload	Off	-	-	-	-
Volume	-20.0	-20.0	20.0	0.1	dB
Mute	Off	-	-	-	-
Signal	Off	-	-	-	-
Level	0	-60.0	20.0	0.1	dB
Overload Thd.	3.0	-20.0	20.0	0.1	dB

❖Element Description

- Overload

  Indicates channel level above chosen Overload Threshold (dB).

- Volume

  Meter showing the channel level.

- Mute

  Mutes or un-mutes the output channel; LED, lit red with channel muted.

- Signal

  To indicate the channel is active or not.

- Level

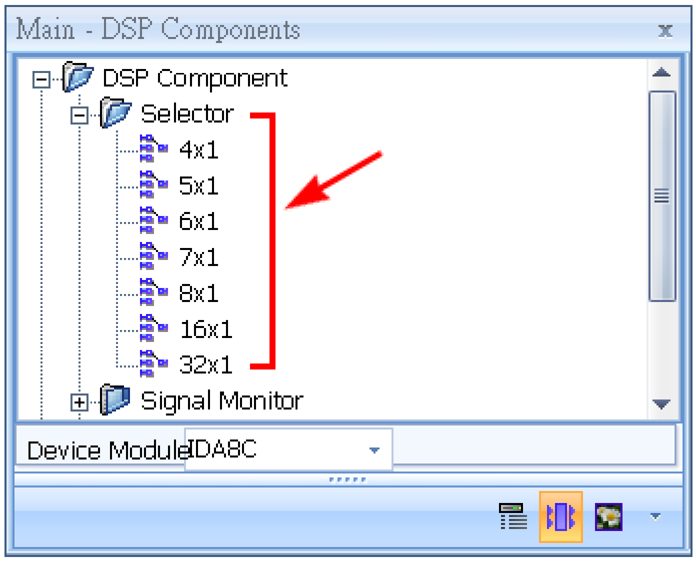
  Signal output level of the module.

- Overload Thd.

  Sets the threshold of the overload indicator.

6.23 Selector

❖Component Template



❖Component Appearance

❖Description

Simple selector allowing to select which input is routed to the single output.

❖Control Window

❖Element Properties

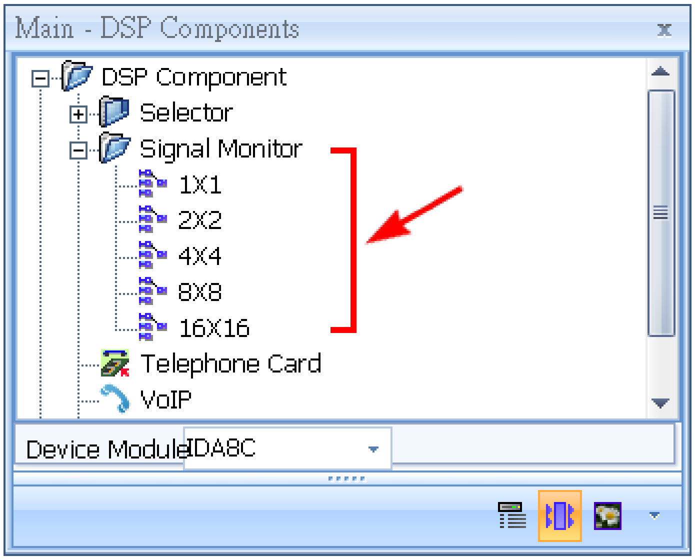
Name	Initial	Minimum	Maximum	Precision	Unit
Input Select	CH1	-	-	-	-

❖Element Description

- Input Select

6.24 Signal Monitor

❖Component Template



❖Component Appearance

❖Description

This component detect the input signal and show the active status of it. If the condition is satisfied element Trigger On will be set to On and logic output channel will be set to hi.

❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Trigger On	Off	-	-	-	-
Threshold	-20.0	-50.0	20.0	0.1	dB
Volume	-40.0	-40.0	20.0	0.1	dB
Response Time	0	0	500	10	ms
Hold Time	300	10	5000	10	ms

❖Element Description

- Trigger On

  Show the input signal is active or not. If the input signal continuously greater than Threshold over Response Time, the trigger on value will be set to 1, and the corresponding logic output is set to hi.

- Threshold

  The threshold to determine audio input is active or not.

- Volume

  The volume of the input signal.

- Response Time

  The time to determine audio input is active or not.

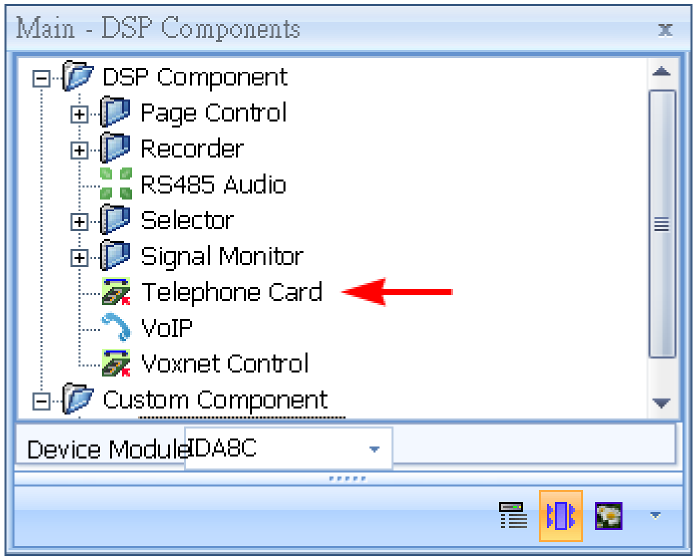
- Hold Time

  The time to determine audio input is active or not.

6.25 Telephone Card

6.25.1 TC Transmit

❖Component Template



❖Component Appearance

❖Description

This component allow Ateis Device to dial or receive phone calls, it need to work with telephone card hardware. The following items are definition of pins on component:

Pins on top side:

Rd: Redial if state of signal from lo to hi.

F: Flash, to make a 3-way calling. active if state of signal from lo to hi.

Pins on bottom side:

Rd: Ready status.

S: Set status

F: Fault status

C: Connect status

D: Dialing status

Ri: Ringing status

B: Busy status

M: MCU status

❖Control Window

❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Dialing	Off	-	-	-	-
Ringing	Off	-	-	-	-
Set	Off	-	-	-	-
Ready	Off	-	-	-	-
Mute	Off	-	-	-	-
Connect	Off	-	-	-	-
Busy	Off	-	-	-	-
MCU	Off	-	-	-	-
Fault	Off	-	-	-	-
Level	0	-15.0	12.0	0.1	dB
Delay Key	None	-	-	-	-
Delay Time	2	1	7	1	Second

❖Element Description

- Dial

This button has two functions:

Dial: Make a phone call, dial the number in Current Number field.

Accept phone call: When there is a phone call ringing, press Dial button can accept it.

- Hang up

This button to hang up the telephone.

- Digit Keys

  Digit keys are the buttons with circled shape on the left part of control window. There are 16 Digit Keys on panel including Numbers "0" to "9", symbols "*" and "#", Alphabets from "A" to "D".

  If status is ready, The digits be pressed will be recorded in Current Number.

  When press digit key during a phone call session, This component send DTMF signal to remote.

- Current Number

  Recode the number dialed or display incoming calls.

- Back

  Erase the last digit of current Number.

- Last Number

  The latest number dialed.

- Redial

  Redial the Last Number.

- Dialing

  Indicates the telephone card is dialing a number.

- Ringing

  Indicates there is an incoming call need to accept.

- Set

  This LED light up if the external telephone which connect to the telephone card is off hook.

- Ready

  Indicate the status of telephone card, This led light up if the telephone line is ready to make a call.

- Connect

  Indicate the telephone card is in the session of call.

- Busy

  This LED shows the remote telephone is in busy state, and not allow to accept any other calls.

- MCU

  Reserve for engineering status display.

- Fault

  To determine there is an error on telephone line or not.

- Mute

  Mute transmitting audio.

- Flash

  This button to make a 3-way calling, During the connection of first remote phone, press this button, then dial the number of second remote, if second remote accept the call, press Flash again.

Level

  Adjust the level of transmitting audio.

- Delay key define

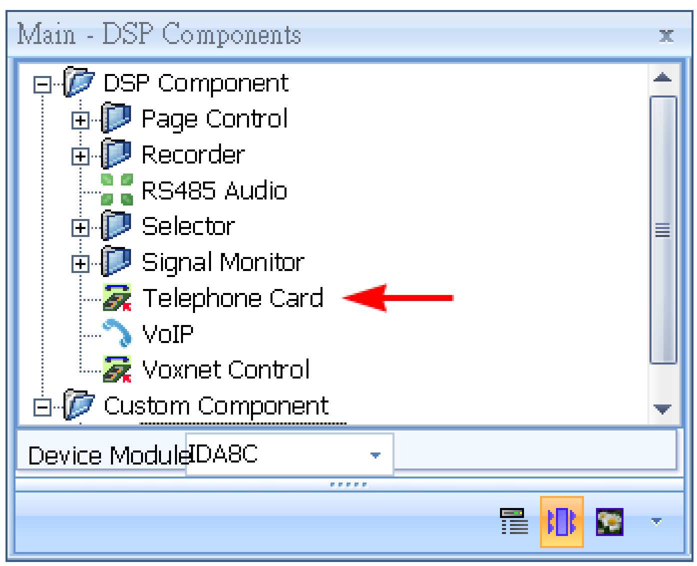
  Define the key for delay. For some telephone systems, it need to dial 0 for an outside line then dial required number. Between 0 and outside phone number, it requires a short delay. In such case, you can define digit "#" for a delay, the number to dial should be "0#XXXXXXXXXX" where X is the outside phone number you want to call.

- Delay Time

  Set the delay time for an delay digit.

6.25.2 TC Receive

❖ Component Template

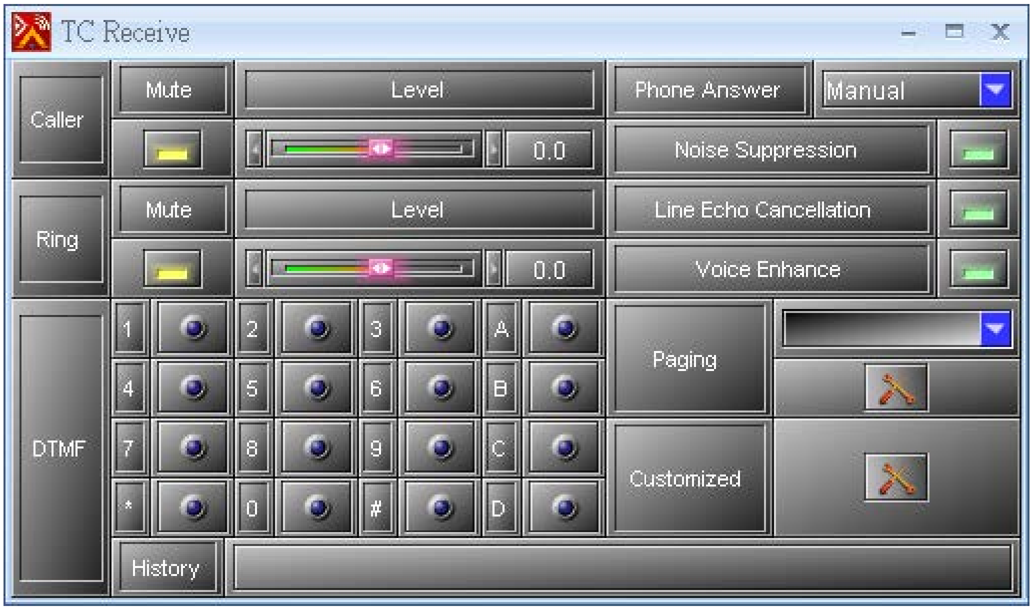


❖ Component Appearance

❖ Description

This component receive telephone call and detect DTMF tone output logic signal response to the key pressing on remote telephone. It need to work with telephone card hardware. The pins on the bottom side of component output the DTMF status.

❖Control Window



❖Element Properties

Name	Initial	Minimum	Maximum	Precision	Unit
Caller Mute	Off	-	-	-	-
Caller Level	0	-15.0	12.0	0.1	dB
Ring Mute	Off	-	-	-	-
Ring Level	0	-15.0	12.0	0.1	dB
Phone Answer	Manual	-	-	-	-
Noise Suppression	On	-	-	-	-
Line Echo Cancellation	On	-	-	-	-
Voice Enhance	On	-	-	-	-

❖Element Description

- Caller Mute

  Mute the audio of caller.

- Caller Level

  Adjust audio level of caller.

- Ring Mute

  Mute ring audio.

- Ring Level

  Adjust level of ring.

- DTMF

  After an incoming call was accepted, if caller press any key on phone, telephone card will detect and display the status on control window, also change the state of logic output corresponding the number detected.

- History

  Display the DTMF number ever detected.

- Phone Answer

  The mode of action for answering incoming calls. There are two kind of modes:

  - Manual

    Press Dial button on control window of TC Transmit for accepting calls.

  - After N Rings

    Telephone accept calls automatically after rings N times.

- Noise Suppression

  To enable or disable noise suppression function.

- Line Echo Cancellation

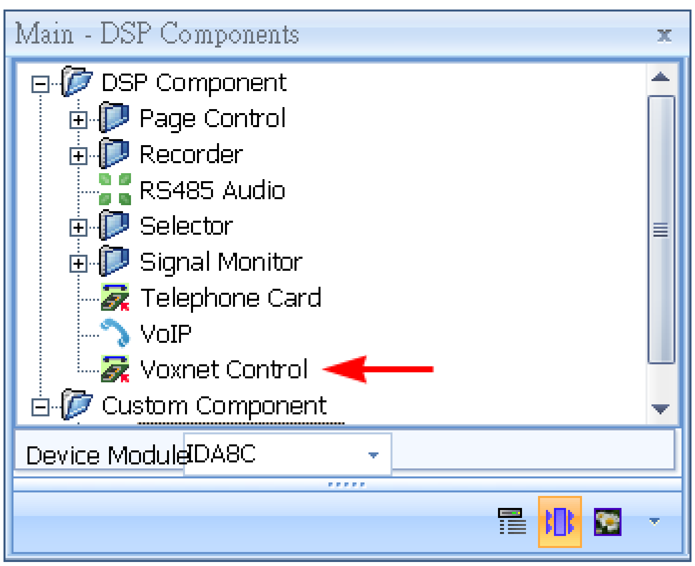
  To enable or disable line echo cancellation function.

- Voice Enhance

  To enable or disable voice enhance function.

6.27 Voxnet Control

❖Component Template

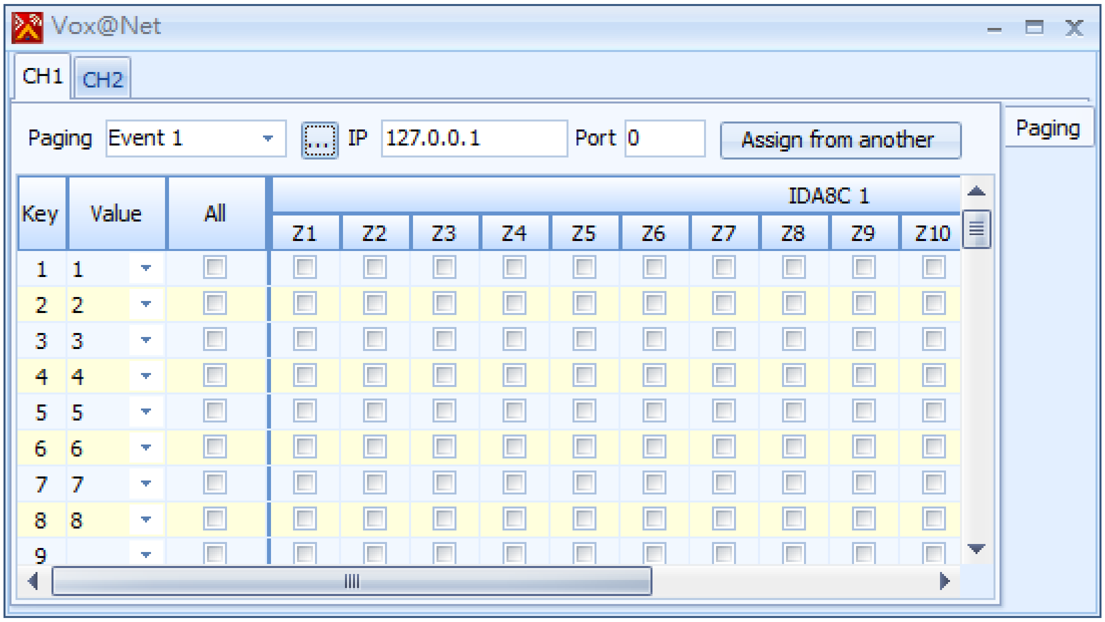


❖Component Appearance

❖Description

This component is an bridge between Ateis device and Voxnet devices. It map the key in voxnet protocol to the code in event for paging control.

❖Settings Window



There are two channels for paging with voxnet source, each source with the following settings:

- Paging Event

  Specify the paging event to bind with the voxnet source for paging.

- Key

  Represent the key of voxnet protocol.

- Value

  To specify the code mapped to the key. each code represent a zone select combination for paging.

- All

  Select all zones.

- Zone Selection

  Click the check box inside the grid on right side of window to select zone to be paged.

7 Contact Infomation

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